AKAI S3200
MIDI STEREO DIGITAL SAMPLER
Software Version 1.0
Operator's Manual

To show our support for the protection of the earth's environment, this manual has been printed entirely on recycled paper.

WARNING!!

To prevent fire or shock hazard, do not expose this appliance to rain or moisture.

CAUTION

RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION : TO REDUCE THE RISK OF ELECTRIC SHOCK,
DO NOT REMOVE COVER (OR BACK).
NO USER-SERVICEABLE PARTS INSIDE.
REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

THE SYMBOLS ARE RULED BY UL STANDARDS (U.S.A)

The lightning flash with the arrowhead symbol superimposed across a graphical representation of a person, within an equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure; that may be of sufficient magnitude to constitute a risk of electric shock.

The exclamation point within an equilateral triangle is intented to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

WARNING

Power requirements for electrical equipment vary from area to area. Please ensure that your S3200 meets the power requirements in your area. If in doubt, consult a qualified electrician or Akai Professional dealer.

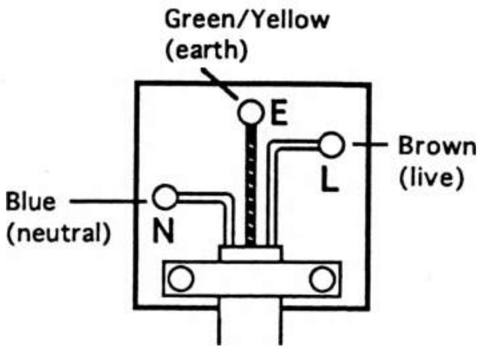
120V	@ 60Hz for USA and Canada
220V-230V	@ 50Hz for Europe (excluding UK)
240V	@ 50Hz for UK and Australia

PROTECTING YOURSELF AND THE S3200

* Never touch the AC plug with wet hands.
* Always disconnect the S3200 from the power supply by pulling on the plug, not the cord.
* Allow only an Akai Professional dealer or qualified professional engineer to repair or reassemble the S3200. Apart from voiding the warranty, unauthorized engineers might touch live internal parts and receive a serious electric shock.
* Do not put, or allow anyone to put any object, especially metal objects, into the S3200.
* Use only a household AC power supply. Never use a DC power supply.
* If water or any other liquid is spilled into or onto the S3200, disconnect the power, and call your dealer.
* Make sure that the unit is well-ventilated, and away from direct sunlight.
* To avoid damage to internal circuitry, as well as the external finish, keep the S3200 away from sources of direct heat (stoves, radiators, etc.).
* Avoid using aerosol insecticides, etc. near the S3200. They may damage the surface, and may ignite.
* Do not use denaturated alcohol, thinner or similar chemicals to clean the S3200. They will damage the finish.
* Make sure that the S3200 is always well-supported when in use (either in a specially-designed equipment rack, or a firm level surface).
* When installing the S3200 in a 19" rack system, always allow 1U of ventilated free space above it to allow for cooling. Make sure that the back of the rack is unobstructed to allow a clear airflow.

UK CUSTOMERS

Important safety notice The flex supplied with this machine has three wires, as shown in the illustration.



*****WARNING: THIS APPLIANCE MUST BE EARTHED*****

IMPORTANT

The wires in this mains lead are coloured in accordance with the following code:

Green and yellow - earth

Blue - neutral

Brown - live

As the colours of the wires in the flex may not correspond to the colour markings in your plug, make sure that wires are connected in the following way.

The green and yellow wire should be connected to the terminal marked 'E' or marked with the safety earth symbol (⏚); the blue wire is connected to the terminal marked 'N', or coloured black. The brown wire should be connected to the terminal marked 'L', or coloured red. Make sure all terminal screws are tightened and there are no loose strands of wire. Ensure also that the flex is securely fastened by the plug's cable grip.

This equipment conforms to No. 82/499/EEC, 87/308 EEC standard

CONFORME AL D.M. 13 APRILE 1989 DIRETTIVA CEE/87/308

FCC WARNING

This equipment has been tested and found to comply with the limits for a Class B digital device pursuant to Part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to correct the interference by one or more of the following measures:

- Reorientate or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

AVIS POUR LES ACHETEURS CANADIENS DU S3200

Le présent appareil numérique n'ément pas des bruits radioélectriques dépassant les limites applicables aux appareils numériques de la Class B prescrites dans le Règlement sur le brouillage radioélectrique édicté par le ministère des Communications du Canada

This digital apparatus does not exceed the Class B limits for radio noise emissions from digital apparatus set out in the Radio Interference Regulations of the Canadian Department of Communications.

FÜR KUNDEN IN DER BUNDESREPUBLIK DEUTSCHLAND

Bescheinigung von AKAI

Hiermit wird bescheinigt, daß das Gerät AKAI
S3200
in Übereinstimmung mit den Bestimmungen der
Amtsblattverfügung 1046/1984
funkentstört ist.

Der Deutschen Bundespostwurde das Inverkehrbringen dieses Gerätes angezeigt und die Berichtigung zur Überprüfung der Serie auf Einhaltung der Bestimmungen eingeräumt.
AKAI ELECTRIC CO., LTD

COPYRIGHT NOTICE

Precautions When Using the Optional MO Disk Drive (SMO-P301) Internally

- **When the SMO-P301 is being used in your S3200, please follow the precautions outlined below.**

1. The MO disk drive has extremely high optical precision and is vulnerable to shock and vibration. If transporting the S3200 by either land or air, we recommend packing it in a shell-shaped case lined with urethane rubber which meets ATA specifications. Never move the S3200 with the MO disk still inside it, as this can cause breakdowns.

2. Make sure the S3200 is used in the horizontal position when it has an MO disk inside it. Do not set it on end or tilt it when using it.

3. The MO disk drive is very sensitive to the surrounding temperature. Avoid extreme temperature changes, and do not use it under temperatures colder than room temperature. If possible, the MO disk drive should be used in an air-conditioned room. If the disk drive is moved from a cold location to a warm one, or if the temperature has been raised suddenly, condensation may form inside the drive. If it is impossible to avoid conditions like these, let the drive sit for at least an hour in the new location before using it.
Using the MO disk drive with condensation still on the inside can cause damage to both the disk drive and to the MO disk.

4. Positioning other equipment so that the cooling fan of the MO disk drive is blocked, or stacking it with other equipment on a rack, can cause the internal temperature to rise. This could cause malfunctioning and should be avoided.

5. The cooling fan has a sponge filter attached to it to keep out dust and other foreign matter. The sponge filter can be removed by turning the four face nuts and taking off the fan guard. It should removed and cleaned periodically.

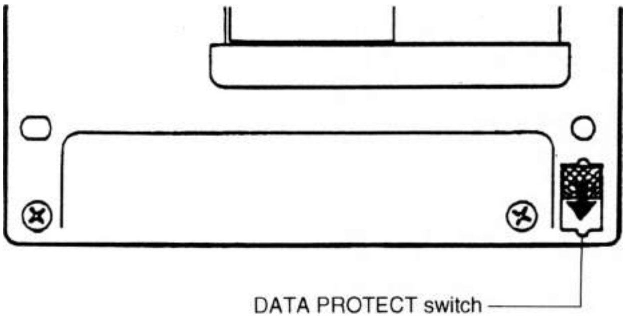
- **About MO Disks**

1. Disks which can be used with the MO disk drive are opto-magnetic (MO) disks which conform to ISO standards and have a 90 mm diameter (3.5 inches). No MO disks are provided with the MO disk drive, so new disks will have to be formatted for use with the S3200 before they can be used. Please refer to page 148 of this operator's manual for instructions on formatting disks.

2. Handling precautions
   - MO disks are cartridge-type disks. Be careful not to drop them or subject them to sharp impact.
   - When the disk cartridge is inserted in the MO disk drive, the shutter opens and the disk is read automatically. Never open the shutter by hand, or touch the inside of the drive.
   - When the disk cartridge is shipped from the factory, the optical precision has already been adjusted. Do not disassemble the cartridge.
   - Never use MO disks in locations where there are sharp differences in temperature, or where there is high humidity. If condensation forms inside the cartridge, it may become impossible to read and write data.
   - Do not insert and eject the MO disk more than necessary.
   - After using the MO disk, always eject it from the MO disk drive.

3. Storing Disks Safely
   - Disks should always be stored in their cases.
   - Never leave disks sitting on the car dashboard or car tray. Also avoid storing disks in the following types of locations:
     * Dusty or dirty locations
     * Where they are exposed to direct sunlight
     * Near a heating device
     * Where there is high humidity

4. Data Protect Switch
The disk cartridge is equipped with a DATA PROTECT switch (a black knob) to prevent data on the MO disk from being erased by mistake. Sliding this switch in the direction indicated by the arrow inhibits writing to the disk.

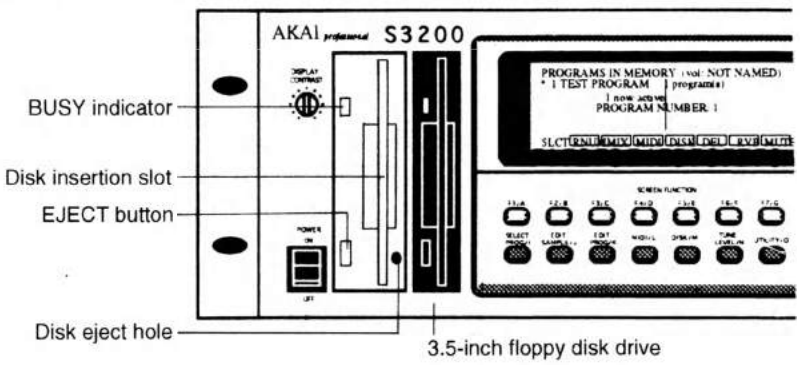


• Cleaning the MO Disk Drive and MO Disks

1. Cleaning the MO disk drive
This MO disk drive uses the optic lens when reading and writing data. If the lens becomes dirty or dusty, it may be impossible to read and write data correctly. We recommend purchasing the special cleaning cartridge used with the MO disk drive (MOA-L33 by Sony) and cleaning the disk drive periodically.

2. Cleaning MO disks
When MO disks have been used over a long period of time, dust and dirt may cling to the disk surface, making it impossible to read and write data correctly. To prevent this problem, the disks should be cleaned once every three months, using a disk cleaning kit.
We recommend using the MOA-D31 Disk Cleaning Kit manufactured by Sony.

• Names and Functions of Parts



BUSY Indicator — This orange indicator lights whenever data is being read from or written to an MO disk. Also, if the internal temperature of the S3200 rises abnormally high, this indicator flashes at 1-second intervals.

EJECT Button — Press this button to eject an MO disk. Never eject disks, however, if the BUSY indicator is lighted (disks may be ejected if the indicator is flashing at 1-second intervals).

Disk Eject Hole — If, for some reason, the MO disk cannot be ejected using the normal procedure, turn off the power to the S3200 and carefully press the manual eject tool provided as an accessory (or a straightened paper clip) into this hole to pop the disk out.

Disk Insertion Slot — MO disks are inserted here. When inserting an MO disk cartridge in the drive, always make sure that the side with the arrow is facing the right, and that the disk is inserted perpendicular to the drive surface (the front panel).

WARRANTY

AKAI Electric Co. Ltd warrants its products, when purchased from an authorized AKAI dealer, to be free from defects in materials and workmanship for a period of 12 (twelve) months from the date of purchase. Warranty service is effective and available to the original purchaser only, and only on completion and return of the AKAI Warranty Registration Card within 14 days of purchase.

Warranty coverage is valid for factory-authorized updates to AKAI instruments and their software, when their installation is performed by an authorized AKAI Service Centre, and a properly completed Warranty Registration has been returned to your Akai Professional dealer.

To obtain service under this warranty, the product must, on discovery of the defect, be properly packed and shipped to the nearest AKAI Service Centre. The party requesting warranty service must provide proof of original ownership and date of purchase of the product.

If the warranty is valid, AKAI will, without charge for parts or labour, either repair or replace the defective part(s). Without a valid warranty, the entire cost of the repair (parts and labour) is the responsibility of the product's owner.

AKAI warrants that it will make all necessary adjustments, repairs and replacements at no cost to the original owner within 12 (twelve) months of the purchase date if:

1 The product fails to perform its specified functions due to failure of one or more of its components.

2 The product fails to perform its specified functions due to defects in workmanship.

3 The product has been maintained and operated by the owner in strict accordance with the written instructions for proper maintenance and use as specified in this Operator's Manual.

Before purchase and use, owners should determine the suitability of the product for their intended use, and the owner assumes all risk and liability whatsoever in connection therewith. AKAI shall not be liable for any injury, loss or damage, direct or consequential, arising out of the use, or inability to use the product.

The warranty provides only those benefits specified, and does not cover defects or repairs needed as a result of acts beyond the control of AKAI, including, but not limited to:

1 Damage caused by abuse, accident or negligence. AKAI will not cover under warranty any original factory disk damaged or destroyed as a result of the owner's mishandling.

2 Damage caused by any tampering, alteration or modification of the product: operating software, mechanical or electronic components.

3 Damage caused by failure to maintain and operate the product in strict accordance with the written instructions for proper maintenance and use as specified in this Operator's Manual.

4 Damage caused by repairs or attempted repairs by unauthorized persons.

5 Damage caused by fire, smoke, falling objects, water or other liquids, or natural events such as rain, floods, earthquakes, lightning, tornadoes, storms, etc.

6 Damage caused by operation on improper voltages.

IMPORTANT NOTE: This warranty becomes void if the product or its software is electronically modified, altered or tampered with in any way.

AKAI shall not be liable for costs involved in packing or preparing the product for shipping, with regard to time, labour or materials, shipping or freight costs, or time and expenses involved in transporting the product to and from an AKAI Authorized Service Centre or Authorized Dealer.

AKAI will not cover under warranty an apparent malfunction that is determined to be user error, or the owner's inability to use the product.

THE DURATION OF ANY OTHER WARRANTIES, WHETHER IMPLIED OR EXPRESS, INCLUDING BUT NOT LIMITED TO THE IMPLIED CONDITION OF MERCHANTABILITY, IS LIMITED TO THE DURATION OF THE EXPRESS WARRANTY HEREIN.

AKAI hereby excludes incidental or consequential damages, including but not limited to:

1 Loss of time

2 Inconvenience

3 Delay in performance of the Warranty

4 The loss of use of the product

5 Commercial loss

6 Breach of any express or implied warranty, including the Implied Warranty of Merchantability, applicable to this product

INTRODUCTION

Congratulations on purchasing the Akai S3200 sampler! The S3200 is a high performance stereo digital sampler with the following features:

* Polyphony: 32 voices
* A-D conversion: 16-bit stereo with 64-times oversampling
* Internal processing: 28-bit accumulation
* D-A conversion: 20-bit with 8-times oversampling (L/R outs)
18-bit with 8-times oversampling (ind outs)
* Sampling rates: 44.1kHz/22.050kHz
* Phase locked stereo sampling and playback
* Internal memory: 8 Megabytes - expandable to 32 Megabytes
(254 programs/255 samples/1,022 'items')
* Sampling times:
  - 8 Meg: 1 minute 29 seconds (mono/44.1kHz)
44.56 seconds (stereo/44.1kHz)
2 minutes 58 seconds (mono/22.050kHz)
1 minute 29 seconds (stereo/22.050kHz)
  - 32 Meg: 5.94 mins (mono/44.1kHz)
2.97 mins (stereo/44.1kHz)
11.88 mins (mono/22.050kHz)
5.94 mins (stereo/22.050kHz)
* Internal effects: Reverb, stereo flanging/chorus, multi-tap delay, delay, pitch shifter
* Inputs: 2 x balanced XLR inputs (L/mono+R)
Stereo balanced jack inputs (L/mono+R)
* Outputs: LEFT/RIGHT outs (2 x balanced XLR and 2 x unbalanced jacks)

  Assignable individual polyphonic outs (8 x unbalanced jacks)

  (Unbalánced outputs use remote ground sensing circuitry to avoid earth loops)
* Stereo Headphone output jack
* Display: 40x8 character LCD
* IB-302D AES/EBU digital interface with real-time digital outputs
* Multi-timbral over 16 MIDI channels

* Internal mixer with variable effects send

* Control of up to 2 x Akai ME35T audio/drum pad to MIDI converters

* Qlist generation for audio/visual post production using IB-303T SMPTE reader/generator

* 12dB/octave resonant lowpass filters

* 2nd bank of 12dB/octave multi-mode filters (lowpass, bandpass, highpass, EQ) with resonance

* Simple tone control for EQ of voices

* 3 Envelope generators (2 multi-stage)

* 2 x Multi-wave Low Frequency Oscillators

* Single triggering legato playback mode

* ASSIGNABLE PROGRAM MODULATION (APM) - The ability to freely assign the following control sources:

  Envelope 1, Envelope 2, Envelope 3, LFO 1, LFO 2, Mod wheel, Pitchbend, Aftertouch, Velocity, Key position, Definable external MIDI controller

  to the following destinations:

  Filter 1 and 2 cutoff frequency, Amplitude, Pan position, Pitch, LFO rate, LFO depth

  in mixable and invertable amounts.

* Editing functions:

  EDIT SAMPLE:

  Trim, Chop, Cut, Extract, 4 Loops, Xfade looping, Auto looping, Join, Merge, Xfade, Gain rescale, Gain normalise, Reverse, Timestretch, Re-sample.

  EDIT PROGRAM:

  APM, resonant lowpass filters, resonant multi-mode filters, 3 envelope generators (2 multi-stage) with envelope templates, 4 way velocity switch/xfade, 2 x LFO's, panning, single trigger legato playback mode, held pitchbend mode, microtonal tuning (with templates)

  EFFECTS: two independent effects sections comprising:

  SECTION 1: REVERB: Type, decay, HF damp, predelay, diffuse, output level, pan, HF cut, stereo spread.

  SECTION 2: CHORUS: rate, depth, feedback, pan, stereo spread, output level, HF cut.

MULTI-TAP DELAY:	delay times, feedback, pan, HF damping, stereo spread, output level, HF cut.
DELAY:	delay time, feedback, LFO rate, LFO depth, pan, output level, HF cut
PITCH SHIFT:	tune, feedback, (independent for left/right), delay, pan, stereo spread, output level, HF cut

* Compatible with S900/S950/S1000/S1100 sound library disks (some library disks may need to be edited, depending on the sound).

* It is possible to record stereo audio directly to a hard disk and this may be played back alongside programs and samples. These recordings may be edited and played back from a Qlist, sequentially in the SONG page or triggered from MIDI.

* A variety of hard disk devices may be used including hard disk, Magneto Optical disks, CD ROM units, Syquest removable cartridges.

* Option to fit 3.5 inch Sony MO 128 Mbyte capacity drive (SMO-P301) or HD108 105Mbyte hard disk drive internally .

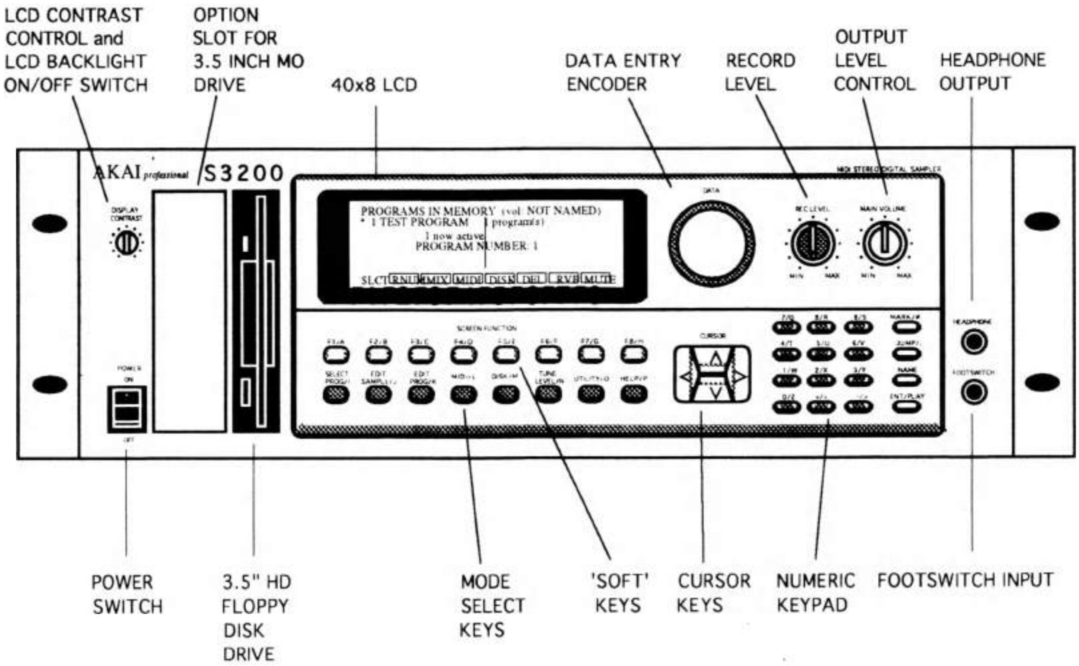
* Back-up to DAT of hard disks and internal memory via the IB-302D digital interface

* HELP pages

* Open software architecture expansion via software updates

The staff at Akai and in particular, the S3200 development team would like to thank you for purchasing the S3200 and hope that you will remain a long time Akai user. The same team who developed the industry standard S1000 and S1100 samplers are responsible for the S3200 and we are confident that your investment will provide you with many years of reliable service.

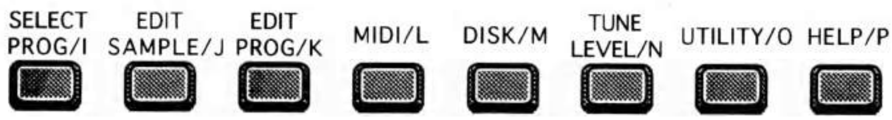
S3200 FRONT PANEL DESCRIPTION



Although the S3200 has a fiarly sparse front panel, behind it is a powerful tool for the creative manipulation of digital audio. However, despite the creative power at your disposal, the S3200 is extremely easy to use.

THE MODE SELECT KEYS

The S3200 is run from 7 illuminated MODE SELECT KEYS that run along the bottom of the control panel. These keys call up different modes of operation which are from left to right:



SELECT PROG/I — In this mode you may select programs for playback. Programs can be selected from the numeric keypad, the DATA control or, of course, via MIDI program change. In this mode you may also set up multi-timbral assignments for sequencing and you may layer programs. You also have access to the S3200's mixer and effects section. A simple disk load function is available for quick loading of data from disk.

EDIT SAMPLE/J — This mode is where you do all your sampling and sample editing. Here you may tune, trim, loop, join, merge, re-sample and timestretch samples.

EDIT PROG/K — This mode is where you assemble your 'raw', edited samples for playback from your MIDI controller. Here

you set key ranges for the samples and you may access the S3200's resonant filters, apply envelopes, modulation, level, tuning, velocity switching and crossfading, microtonal tuning, etc..

MIDI/L

This sets the S3200's 'global' MIDI parameters such as receive channel, program change select, etc.. There are also various diagnostic pages that allow you to view incoming MIDI signals in case of some problem.

DISK/M

This mode is where you may load and save data to and from disk. You may use it to access floppy disks and a variety of hard disk units. Files may also be deleted from disk in this mode.

TUNE LEVEL/N

This mode allows you to tune the sampler and set its operating output level.

UTILITY/O

The UTILITY mode gives you access to the SMPTE and QLIST pages and also to the ME35T programming pages. In this mode, too, you may access the DAT back-up and disk recording functions.

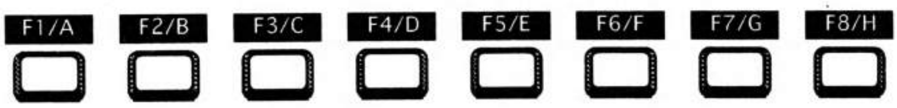
THE HELP KEY

The final key in this row is not a mode select key but a HELP function.

HELP/P

The help screens give you the most relevant section of the manual when the cursor is resting on a particular field. To get help, when the cursor is on a field you don't quite understand, press the HELP button. The help key locks and displays the text on the screen. To discover the function of one of the soft keys, press the HELP key (it will first display information about the current parameter) and then press the particular function key you are interested in. Its function will be described on the LCD. Pressing the HELP key again will turn the function off.

THE SOFT KEYS



The SOFT KEYS directly under the LCD call up various functions and pages within each mode - these vary from mode to mode and have no pre-defined function. As such, they cannot be easily explained here! There are many common keys in many of the functions, however, such as, for example, COPY, RENAME and DELETE which are always on F6, F7 and F8 in those pages where they appear. Commands such as GO, and ABORT always appear on F7 and F8. There is also a convention to the type of functions available which is worth noting.

If a key has this highlighted type of display: REC1 then this signifies that pressing this key will take you to another page.

If the key has this type of display: COPY (i.e. the function simply has a box around it and is unhighlighted) then this signifies that the key is an 'action' key and will instigate some kind action such as SAVE, LOAD, DELETE, GO, ABORT, etc..

If the key has no form of box around it and is not highlighted - i.e: REC2, then this indicates the page you are currently in although this is always shown in the top left corner of the display as well. This type of key switches between two types of display - pressing it once displays note names, pressing it again displays notes as MIDI note numbers. In EDIT SAMPLE, you may toggle between a sample point and millisecond display.

If you are unsure of the function of any soft key, please use the HELP pages by pressing the HELP key followed by the soft key you are interested in.

SELECTING PARAMETERS AND ENTERING DATA

You move around the screen using the CURSOR KEYS and data is input either from the DATA ENTRY ENCODER or from the NUMERIC KEYPAD. You may move around within digit fields using the +/< and -/> keys found on the numeric keypad.

On simple fields like filter cutoff, attack time, MIDI channel, etc., that have two digits, you may simply type in a two digit number - i.e. 23, 45, etc.. On such fields, you will find that the DATA ENTRY ENCODER will cover the whole range quite quickly so you may find that more convenient. The same is true of three digit fields. Turning it clockwise increases numeric values, and turning it counter-clockwise decreases these values. You may also use the +/< and -/> keys to position the cursor on the 'tens' or 'hundreds' field to make more rapid changes. For non-numeric values (i.e. sample rates, sample type, loop type, etc.), turning the encoder will display all the options in order. Normally, there is no other entry procedure required; simply displaying the correct value of a parameter using the DATA encoder selects and stores it into the S3200's memory. Our sound programmers at Akai usually use a combination of the numeric keypad and the DATA ENTRY ENCODER for speedy input of parameter values.

When editing numeric parameters, some of the values can be quite large, and it would be necessary to turn the DATA ENTRY ENCODER thousands of times (literally!) in order to go through the whole range if the value was only changed by 1 for every click of the DATA control. There is an alternative to turning the control thousands of times, though. When you press the CURSOR keys, you move from one parameter to another and, using the +/< and -/> keys, you may move around within a large numeric field.

If a number such as 123456.78 is displayed, and the +/< key is pressed so that only the first three digits are highlighted thus:

123456.78

turning the DATA ENTRY ENCODER clockwise by one step now will increase the value of the last highlighted digit, so:

124456.78

Now if the -/> key is pressed once, the first four digits will be highlighted:

124456.78

and turning the DATA ENTRY ENCODER one click clockwise will produce:

12456.78

If you turn the DATA control more than ten clicks, of course, the value of the whole parameter will be incremented or decremented by the number of clicks. In this way, with very little effort, fast accurate editing of numbers can be achieved using only the +/< and -/> keys and the DATA ENTRY ENCODER. The best way to learn how this works is to practise; after a short time, it should become second nature.

On 'signed' fields - that is, fields that have a + or a - before them, the +/< and -/> will two things. Pressing the +/< key will move the cursor left within the field and, when it reaches the furthermost left digit, you may use it to switch between + and - depending on the selection you wish to make. The -/> key will move the cursor right and, when it reaches the furthermost right digit you may toggle between + and - again.

As mentioned above, as an alternative to turning the DATA control, you can also use the numeric keypad for direct entry of data. When you know the exact number you want to enter, this can be faster than using the DATA control, but when experimenting (for example, when setting loop points or sample start and end times), the DATA control may be faster than the numeric keypad. You'll probably discover quickly what method works best for you in each situation.

***NOTE**: The cursor always stays on the last currently selected field in any screen. For example, if you are in, say, the filter pages and are setting envelope 2's depth and then go to the ENV2 page to make an adjustment there, when you return to the filter page, the cursor will still be on envelope 2's depth parameter.*

The other remaining front panel keys are found to the right of the numeric keypad and are:

MARK AND JUMP KEYS

These two grey buttons to the right of the number keypad are used in conjunction with each other. If you are carrying out editing operations which require changing display pages a lot, these can save a lot of time and effort. Pressing the MARK/# button when the cursor is on a field will cause the S3200 to remember the position of the cursor, and pressing JUMP/. will take the cursor back to the MARK(ed) position from any other page. Pressing JUMP/. again will take you back to the page and function you were at before you JUMP(ed).

You can reset the MARK position at any time. This position is lost when the power is turned off. At power-on, this position defaults to the program select field in the initial SELECT PROG page.

NAMING FILES - THE NAME KEY

When samples, programs, effects or drum input settings are changed, they should be given a name for easy reference. Pressing the NAME button in certain pages will enable you to name the data and you will notice that each button has a letter following its primary function (i.e. EDIT PROG/K, F4/D, HELP/P). Up to 12 characters (uppercase only) are entered by pressing the front panel buttons (although you may also scroll through letters and

numbers using the DATA control). When actually entering names, pressing the NAME button will switch the function of the numeric keypad between letters and numbers. The CURSOR keys moves the cursor around inside the name field when naming a sample or program.

When entering names in letter mode, the +/< and -/> buttons work as backspace and space-bar buttons respectively (when in number mode, they enter the "+" and "-" characters), and the MARK/# and JUMP/. keys enter "#" and "." respectively. The last button, ENT/PLAY, enters and confirms the name and the S3200 prompts you to either copy or rename the item.

THE ENT/PLAY KEY

This is a dual-purpose button. When naming samples, programs, etc, pressing this button will end the naming process in conjunction with COPY and REN(ame) as described above. In other modes of operation, this key will play the sound at a pitch, velocity and MIDI channel as set in the MIDI TRAN(smit) page in the MIDI mode. The default may be freely set as you wish.

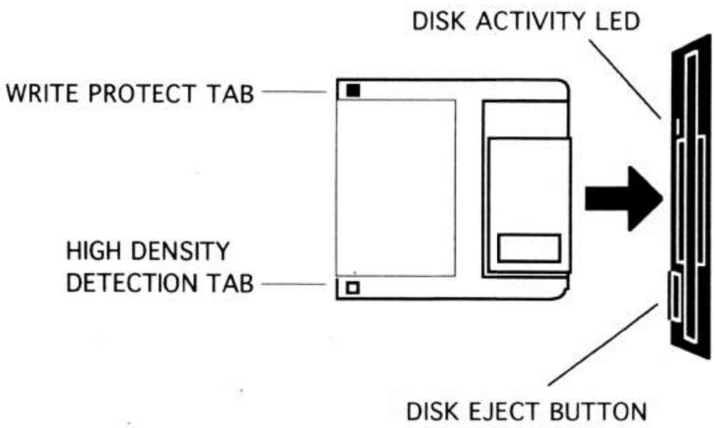
***NOTE**: When editing samples in EDIT SAMPLE, pressing this button will play back the sample at its correct pitch and not necessarily the one set in the MIDI TRAN(smit) page).*

This key is also used in certain modes of the disk recording functions to play back takes from disk.

THE DISK DRIVE

The 3.5 inch floppy disk drive will accept high density and low density disks .

Disks are inserted into the drive thus:



The label should be facing towards the screen and panel controls when it is inserted (actually, it is physically impossible to insert disks the wrong way round without using an extreme amount of brute force!).

To eject the disk, simply press the DISK EJECT button. When a disk is loading, saving or formatting, the DISK ACTIVITY LED will be lit.

***IMPORTANT NOTE**: On the S1000 and S1100, it is possible to format low density (DD or double density) disks to a high density format. This is not possible on the S3200. Furthermore, the S3200 cannot read DD disks that have been formatted on an S1000/1100 to a high density format. You will need to first load these sounds into an S1000/1100, resave them onto high density disks (formatted to high density, of course) before they can be used in the S3200. High density disks have a hole on the right hand side which is used by the disk drive to detect that it is a high density disk. If the S3200 does not 'see' this hole, it assumes it is a DD disk and so searches for a DD format. If it doesn't find it (i.e. because the disk is high density format), it cannot read it.*

It is important to remember that, unlike a synthesizer, the S3200 has no means of storing sounds in an internal memory. The amount of data involved in audio samples would make the cost of battery backed up RAM prohibitively expensive. As a result, it is vital that you save your work to disk before turning the power off otherwise you will lose your work and, unless previously saved or backed up, it will be gone for ever. In fact, it is a good idea to regularly save your work as you are working. All good computer users do this and it prevents the accidental loss of data should power be accidentally removed from the instrument. This also serves as a form of 'undo' - if you make some kind of mistake in your programming and editing and can't fix it, you can load the last level of editing back into the sampler. It may be a bit tedious to keep stopping every now and then to save your work but it is better than losing some valuable sounds.

TAKING CARE OF YOUR DISKS

These floppy disks contain valuable sound data and, as such, should be treated with extreme care. Please observe the following points, therefore:

1 Never slide the metal cover back and touch the disk. Finger marks may render the disk unreadable.

2 Don't leave the disk in the drive wherever possible. When the disk is in the drive, the metal protective cover slides back exposing the actual disk inside - this makes the disk susceptible to picking up dust which may cause read errors.

3 Do not leave your disks in a hot car.

4 Do not place your disks next to any magnetic sources such as speakers, amplifiers, televisions, etc.. Also, try to avoid X-ray machines. At airports, it is sometimes possible to ask for your disks to be inspected by hand at security desks but, with the added security at airports these days, this may not be possible. Always check with the security officer though, just in case. Security X-ray machines are generally safe with disks, though. If in doubt, make backup copies which should be left at home.

***NOTE**: Some checked in luggage is X-rayed by quite powerful machines that are not as safe as those that check hand luggage. It is probably best to take your disks as hand luggage.*

5 Do not leave your disks around when drinking liquids - one accidental spillage could ruin a lot of work!

6 Always use high quality disks. Whilst cheap ones may be appealing, they are prone to errors more than good ones.

7 Try to ensure that the write protect tab is switched on (i.e. the tab blocks the hole). This will prevent accidental erasure, formatting and loss of data. It may be a nuisance to try to write to the disk and find it write protected but it is less of a nuisance than accidentally over-writing a set of your favourite samples and programs!

8 Try to get into the habit of labelling your disks - it will pay dividends in the end when you are searching for something.

9 Invest in a sturdy carrying case for your floppies especially if you are a gigging musician. Heavy duty metal camera cases are ideal and some flight case manufacturers now make special heavy duty disk flightcases.

10 Even if you are using a hard disk of any sort, please make sure you have backed up your work to floppy disks. It can be time consuming but it will be worth it if you ever have a problem with your hard disk!

AND FINALLY...

LCD CONTRAST

You may adjust the viewing angle for the screen using the DISPLAY CONTRAST control.

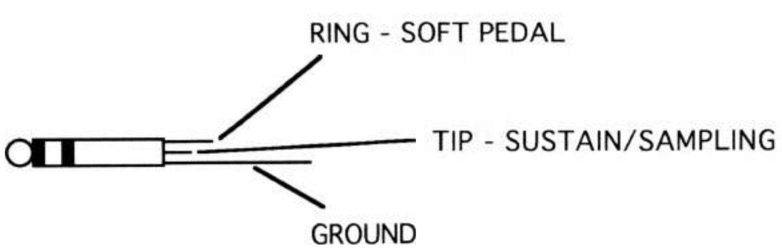
NOTE*: To preserve the life of your LCD, this control also has a switch which you can use to turn the backlight of the LCD on or off. At times when you are not actively using the S3200's front panel for programming (i.e. when actually sequencing or recording to tape, for example, or when using the sampler live on stage), you might like to switch this off. All LCD's of this type progressively get dimmer with age and this switch can help prevent this. The switch should be pushed in to switch the backlight off and pushed in again to switch it back on.*

RECORD LEVEL, MAIN OUTPUT LEVEL, HEADPHONE OUTPUT

Input level for sampling is regulated using the RECORD LEVEL CONTROL and the S3200's overall output level is controlled, not surprisingly, by the MAIN VOLUME control. This also governs the level of the sound appearing at the HEADPHONE OUTPUT.

FOOTSWITCH INPUT

The FOOTSWITCH input is actually two switch inputs using a stereo jack. One input is used for sustain and for initiating sampling (see later) and the other is used for the soft pedal (MIDI controller 67). The wiring of the plug is as follows:

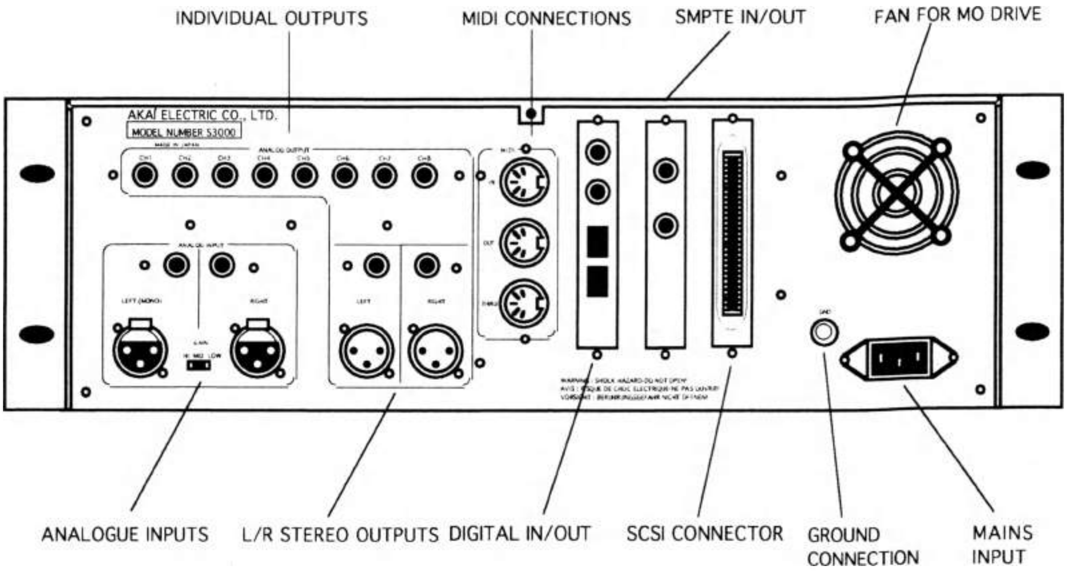


Both require a press-to-open type of switch.

If you have used or owned an S1000 or S1100, the biggest difference you will probably find is the use of cursor keys as a method of getting around the screen. As a result of our success with the Akai DD1000 Magneto Optical Disk Recorder/Editor, these have proved to be a very convenient way of moving the cursor around the screen and so replace the cursor knob found on the older samplers.

THE REAR PANEL

The rear panel consists mainly of input and output connectors.



ANALOGUE OUTPUT CONNECTIONS

There are ten audio connectors provided - a LEFT/RIGHT pair of outputs and eight individual outputs. The simplest way to connect the S3200 to a mixer or amplification system is to use the LEFT/MONO and RIGHT stereo connectors (if you want to make mono connections, use the LEFT/MONO connector only). There are two LEFT/RIGHT connections available - one set is a pair of unbalanced jack sockets, the other a pair of balanced XLR connectors.

Use of the INDIVIDUAL OUTPUTS (CH1 through CH8) allows much more flexibility and control (but of course, this will take up more input channels on the mixing console). Programs can be assigned to any one of these output channels (true stereo programs using stereo samples should be assigned to the stereo outputs for the full stereo effect) and effected separately.

The S3200 uses remote ground sensing circuitry so that the unbalanced outputs are protected against the possibility of ground or earth loops that can sometimes occur in complicated setups where a lot of equipment is connected.

ANALOGUE INPUTS

Two parallel pairs of stereo balanced connectors are provided for connection of sound sources (wired in accordance with American standards - 1-shield, 2-cold, 3-hot), and the other pair is a pair of balanced jack connectors using stereo jack sockets. Unbalanced sources can, of course, be connected to the XLR inputs. If a mono source is used for sampling, use only either the LEFT (MONO) XLR or phone connector.

> ***NOTE**: The XLR connector and phone jack on the analogue inputs are connected in parallel. When inputting an analogue signal, use only one pair of them.*

GAIN SWITCH

This is a 3-position slider switch (LOW, MID, HIGH) used for matching the level of the input source to the recording amplifier of the S3200. Fine adjustment should be carried out with the REC LEVEL control on the front panel. Ideally, you should set the REC GAIN so that the REC LEVEL is set about 2 o'clock. Remember that unlike analogue equipment, digital devices produce distortion which is particularly unpleasant, and “soft clipping” and the effect of saturation found in analogue recordings cannot be obtained when recording digitally. You should always allow sufficient headroom for transient peaks when making a sample.

Note also, that recording at too low a level will not allow you to make full use of the S3200's dynamic range and signal to noise figures.

***NOTE 1**: When making a sample, you may not immediately notice any clipping that may have resulted from incorrect level settings and it may only become apparent when playing back samples lower than the original pitch at which they were sampled.*

***NOTE 2**: If you set the level too low, the S3200's EDIT SAMPLE pages have a GAIN NORMALISE function that allows you to restore the sample to its optimum level for full use of the sampler's wide dynamic range.*

The REC GAIN sensitivities are HI -58dBm, MID -38dBm, LOW - 18dBm.

MIDI IN, OUT, THRU

These MIDI connectors conform to the usual MIDI standard. IN is used to receive MIDI from your keyboard, sequencer or audio/MIDI trigger unit or alternative MIDI controller such as the Akai EWI wind synthesizer as well as for accepting System Exclusive data. MIDI OUT is used for transmitting Note On/Note Off and performance (pitch bend, aftertouch, etc) data, as well as for System Exclusive communication. MIDI THRU echoes the data received at the MIDI IN terminal and is the connection usually used when 'chaining' several pieces of MIDI equipment together.

SCSI CONNECTION

This connection is used to connect an external SCSI hard disk or M.O. disk.

DIGITAL IN/OUT CONNECTION

These connectors allow direct digital connection to other digital audio equipment fitted with similar connections (AES/EBU, coaxial, optical). The connectors are 6.35 mm stereo (balanced) phone jacks and optical jacks (Tosslink®), so you may need to modify your cables.

SMPTE/EBU TIMECODE IN/OUT

These connectors are used to input and output SMPTE/EBU timecode when making a Qlist to timecode. The connectors are 6.35 mm stereo (balanced) phone jacks.

GROUND CONNECTION

This may be used to ground equipment to overcome problems regarding earth loops. This is particularly a problem when rack mounting lots of equipment together.

COOLING FAN

This cooling fan is installed when the optional internal 3.5 inch M.O. disk drive is installed.

POWER CONNECTION

This is used to connect AC power to the S3200.

BEFORE CONNECTING ANY AC POWER, PLEASE ENSURE THAT YOUR UNIT IS DESIGNED FOR YOUR AREA'S POWER SUPPLY. A MISTAKE NOW COULD RUIN YOUR WHOLE DAY NOT TO MENTION YOUR S3200!!!

SETTING UP THE S3200

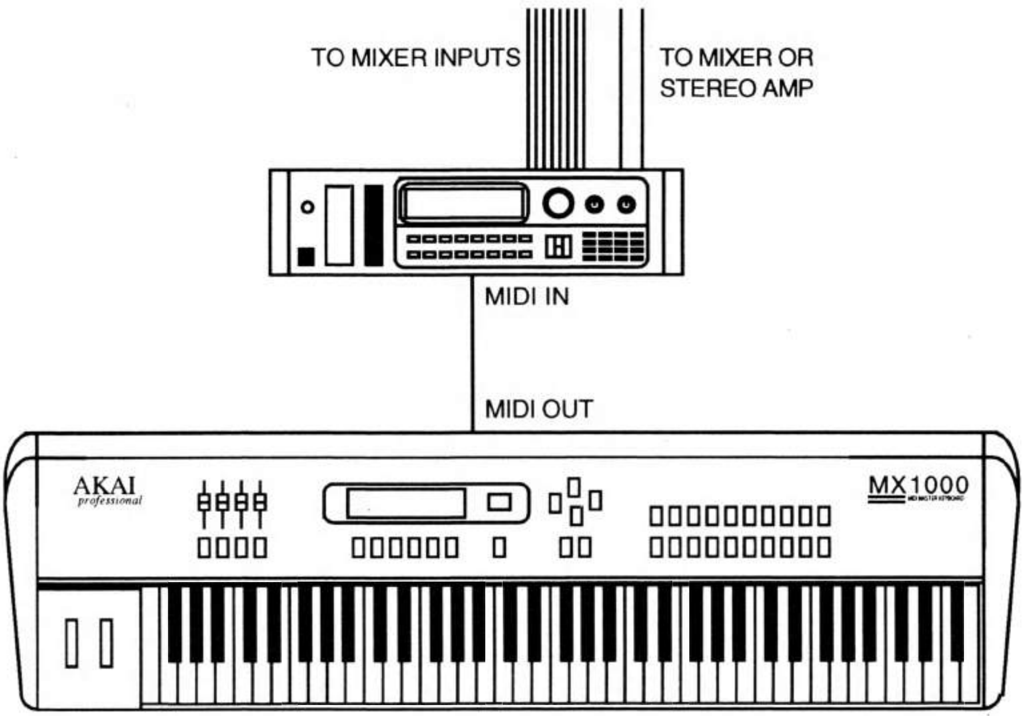
This short section tells you how to get the S3200 'up and running' fast. For full details of other operations, refer to the appropriate explanation in this manual.

CONNECTIONS

Start by placing the S3200 on a firm level surface or in a 19" equipment rack. Remember to leave adequate space for airflow above and behind the S3200 if putting it in a rack (a 1U space is adequate).

NOTE: *If an optional MO disk drive is installed, do not put other rack-mount units directly above and below the S3200. Doing so may cause troublesome heat build-up inside.*

For now, you'll probably want to connect a MIDI controller, such as the AKAI MX1000 master keyboard. Connect a MIDI OUT of the controller to the S3200's MIDI IN connector. The power-on default of the S3200 is MIDI channel 1 so please set your MIDI controller to match that. However, unless you want to play through headphones, make some audio connections. Using the LEFT/MONO and RIGHT stereo outputs will probably be sufficient at this stage, so connect these to two channels of a mixing console, and set the gain appropriately (the standard output level is 7.5dBm and the impedance is 600 ohms). If you are connecting the LEFT/RIGHT outputs to a mixer, set the pan pots for the mixer channels hard right and hard left so that you get the full benefit of the stereo capabilities of the S3200. Connect the power lead to the AC supply and to the S3200.



GETTING STARTED

You are presumably now anxious to use your new acquisition so enough of this manual reading for the moment!!

Assuming there is some form of MIDI controller connected to the S3200's MIDI input and assuming you have not yet switched the instrument on, insert one of the library disks that come with the S3200 into the 3.5" floppy disk drive (if you really are new to samplers, please have a quick look at the section INSERTING THE FLOPPY DISK in the introduction).

With the disk in the drive, switch the S3200 on - it will automatically load the disk on power up.

You will see some activity going on as the S3200 loads the sounds (the disk activity LED will light and a quick look at the LCD will show a "LOADING...." message). After about 40-50 seconds, this will stop - you may now play the S3200.

The library disks supplied each come with several programs. These either use different combinations of samples or a variations on a set of samples. These may be selected from the S3200's front panel using the DATA encoder or alternatively, you may use MIDI PROGRAM CHANGE commands from your MIDI controller to select new programs.

Once you have tired of the first disk you loaded, take another and insert it into the disk drive. Press DISK (F5) and VOL (F8) to load it. After 40-50 seconds, you should be able to play the sounds on that disk. Again, use the DATA encoder or MIDI program change to select the different programs. Repeat the process for the other disks.

If, at this point, you want more sounds, aside from making them yourself (an easy enough process as we shall see), please contact your dealer or Akai distributor who will be able to advise you from where and how you may obtain Akai or third party sound library. Of course, you may already have other sound library disks if you are an S1000 or S1100 user in which case you are probably not reading this section so why are we wasting time explaining this - you already know what to do!!!

HOW THE S3200 WORKS

Despite its versatility, the S3200 is very straightforward and once you have a basic grasp of the flowcharts shown on the next pages, things will make more sense.

Basically, you can have SAMPLES. These are the pieces of raw digital audio that are always the basis of any sound in the S3200. These may be derived from floppy disk, hard disk (including Magneto Optical (MO) Disks, removable cartridge types, etc.), CD ROM or, of course, you may sample your own sounds via the analogue or digital inputs.

Once you have a raw sample, there are many things you can do with it within EDIT SAMPLE. You may TRIM it - that is, remove any unwanted audio from the start or end. You may CHOP it - that is, remove a section in the middle and splice the two remaining sections together or you may CUT it - that is, remove a section in the middle and keep the gap thus created. You may also EXTRACT part of the sound - that is isolate a section (such as one snare drum in a breakbeat) and extract that snare hit from the original. You may also TUNE and/or REVERSE the sample.

Of course, one problem always associated with sampling is LOOPING. Because any given sample is only a few seconds long, if you want to sustain that sample longer than its original length, some method has to be sought to do this. This is LOOPING. A loop is a section of the sound that repeats as long as you hold your finger on the key and is set by setting a start and end point for the loop. This is something of an art and a science and something we won't delve into here in great detail. To make looping as easy as possible, the S3200 has FIND and CROSSFADE functions to help you set good loops.

Other sample editing functions include TIMESTRETCH which allows you to lengthen or shorten a sample without changing its pitch (again, we will look at this in more detail later in this manual) and RE-SAMPLING, a technique that allows you to squeeze the optimum performance out of available memory space. On the S3200, it is also possible to JOIN samples end to end to create long, evolving sounds or you may MIX samples to create big layered textures. You may also NORMALISE and RESCALE a sample's level for optimum signal to noise performance.

Once you have edited a sample you may place it into a PROGRAM.

A PROGRAM is where you assemble your sample(s) for playback. In EDIT PROG, you allocate a sample to a KEYGROUP (in fact, you can allocate 4 samples to one keygroup for layering, velocity switching and velocity crossfading but more on that later!). A program may have as little as one keygroup spanning the entire keyboard or as many as one keygroup for every key each with four samples in them! To overcome the abrupt tonal discrepancies sometimes experienced when two different samples are placed 'side by side' on the keyboard, positional keygroup crossfading can be used for a smoother transition.

Once in a keygroup, the sample may be passed through resonant filters for tonal modification, through a second bank of multi-mode filters and a simple EQ section for further tonal refinement and special effects, through amplifiers for amplitude control and through a pan section for stereo placement. Keygroups may be freely assigned to the individual outputs for external processing on a mixing console.

On top of this, you may apply modulation from Low Frequency Oscillators (LFOs) for vibrato and other effects. A SINGLE TRIGGER LEGATO mode allows you to emulate solo instruments such as woodwind and brass more realistically and to allow for greater expression when playing synth bass parts or lead sounds.

One powerful feature of the S3200 is the flexible ASSIGNABLE PROGRAM MODULATION known as APM for short) in EDIT PROGRAM. Using this, any modulation source may be routed to virtually any control input. In this way, the S3200 is a powerful synthesizer as well, except that you may use almost any sampled sound (of your own or from a sound library) as the basis of your own powerful and expressive sounds.

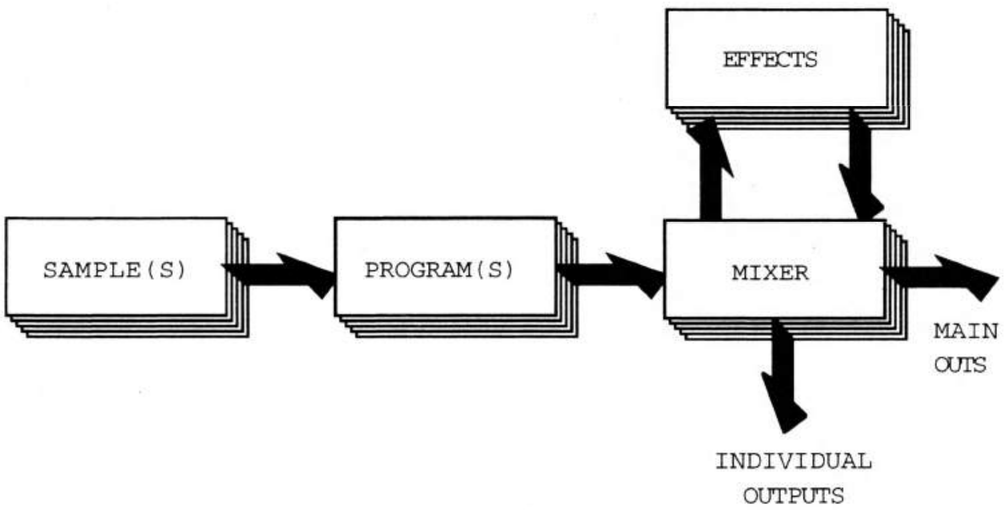
Once you have placed your sample(s) into a program, you may play them from a MIDI controller. Programs may be sent in varying amounts to the S3200's internal effects and programs may also be layered or set to different MIDI channels for multi-timbral sequencing. In this application, the S3200's built-in mixer allows you to balance the various sounds together without needing to use up channels on your mixer (although the individual outputs do allow for external processing as well, don't forget).

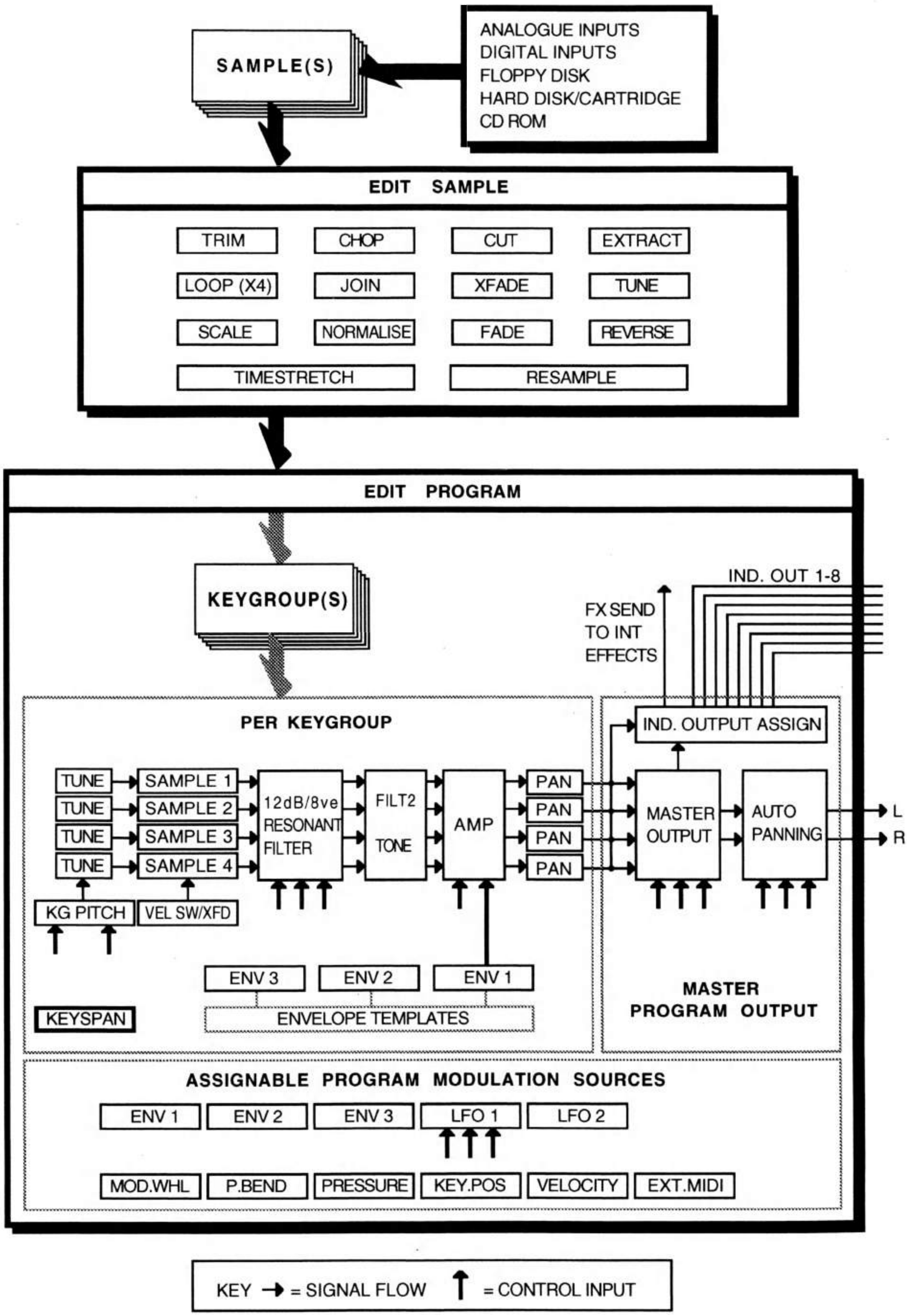
Once you are happy with that, the whole thing may be saved to disk for recall at a later date.

One of the beauties of the Akai samplers is that there are no restrictions on what you do with samples - any sample can be placed into any keygroup in any program. One sample may be placed into a variety of different programs, each of which impose different envelope or filter settings to create a wide range of possibilities from very little. One or more samples may be used in any other programs even on other disks and/or one program can be used as the basis for processing other samples.

S3200 FLOWCHART

The following flowcharts will help your understanding of the S3200.




SAMPLE(S)
ANALOGUE INPUTS
DIGITAL INPUTS
FLOPPY DISK
HARD DISK/CARTRIDGE
CD ROM
EDIT SAMPLE
TRIM
CHOP
CUT
EXTRACT
LOOP (X4)
JOIN
XFADE
TUNE
SCALE
NORMALISE
FADE
REVERSE
TIMESTRETCH
RESAMPLE
EDIT PROGRAM
KEYGROUP(S)
FX SEND TO INT EFFECTS
IND. OUT 1-8
PER KEYGROUP
TUNE
SAMPLE 1
TUNE
SAMPLE 2
TUNE
SAMPLE 3
TUNE
SAMPLE 4
12dB/8ve RESONANT FILTER
FILT2
TONE
AMP
PAN
PAN
PAN
PAN
IND. OUTPUT ASSIGN
MASTER OUTPUT
AUTO PANNING
L
R
KG PITCH
VEL SW/XFD
ENV 3
ENV 2
ENV 1
ENVELOPE TEMPLATES
KEYSPAN
MASTER PROGRAM OUTPUT
ASSIGNABLE PROGRAM MODULATION SOURCES
ENV 1
ENV 2
ENV 3
LFO 1
LFO 2
MOD.WHL
P.BEND
PRESSURE
KEY.POS
VELOCITY
EXT.MIDI
KEY = SIGNAL FLOW = CONTROL INPUT

SELECT PROG

```
PROGRAMS IN MEMORY (vol: VOLUME 020     )
*  1 STRINGS 1      |  8 program(s)
   2 PIANO          |  1 now active
   3 BASS           | PROGRAM NUMBER:  1
   4 BIG KIT        |
   5 SYN BASS       |
 SLCT RNUM  MIX  MIDI DISK  DEL  RVB  MUTE
```

This display shows the main SELECT PROG screen and here you may select and play programs. You may also set up sophisticated multi-timbral arrangements of programs for sequencing from MIDI and you may also perform layering and keyboard split techniques. Within this mode is a mixer for setting the levels, pan position and effects send of programs and here you may also access and program the effects section. You also have access to the disk operations for loading sounds.

SELECTING PROGRAMS

If you have read the chapter GETTING STARTED, then you may already be familiar with selecting sounds on the S3200. This can be done in several ways:

1 In the main SELECT PROG screen as shown above, turning the DATA control will select different programs for playback.

2 In the main SELECT PROG screen, use the numeric keypad to select sounds by number. To do this, press the </+ key twice followed by a three digit number. I.e. to select program 25, press </+ and 025. To then select program 26, simply press 6. To then select program number 1, press </+ once followed by 01 or twice followed by 001.

3 Send MIDI program change commands from your keyboard, synthesizer or whatever MIDI controller you are using. You will note that the S3200 uses 'actual' program numbers 1-128 - if your keyboard sends out 0-127, please bear this in mind as you will have to offset all program change commends you issue by 1.

Whichever method you use for selecting programs, when the program is selected, an asterisk appears alongside it number and the program number is indicated. One useful feature of the S3200 is that it is possible to be playing one program whilst another is being selected. For example, whilst holding down a low string note you could select, say, a brass sound. The strings will continue to sound and you may now play the brass part. Please note, however, that if you are using the internal effects on either or both of these sounds, there may be a noticeable change as only one effect can be used at once. In this case, the brass program's effects would take priority.

VIEWING PROGRAMS

The LCD can display 5 programs at any one time. There may be occasions when you have more than this and so you may use the CURSOR keys to scroll through your program list. Pressing the CURSOR LEFT or RIGHT keys will scroll though the list one at a time and pressing the CURSOR DOWN or UP keys will increment through the list in steps of five programs. This can be helpful if you are searching for a particular sound and need to see what is

loaded in the S3200. The display will tell you how many programs are loaded and in the above example, we can see that there are actually 8 programs in memory even though only the first five are displayed. It is possible to be viewing other programs whilst playing another.

RENUMBERING PROGRAMS

On the S3200, program numbers correspond to patch numbers on a synthesizer. When a MIDI Program Change message is received, the appropriate program is selected. However, to match S3200 programs with the patch numbers on your synthesizer, you may want to re-number the programs, so that selecting a brass sound on the S3200 will call up a similar (or complementary) patch on a remote synthesizer module. To do this, press the RNUM key. You will receive something like this screen display:

```
CHANGE PROGRAM NUMBER OF MEMORY PROGS.
* 1 STRINGS 1
  2 PIANO
  3 BASS
  4 BIG KIT
  5 SYN BASS
SLCT RNUM                      ALL  SLIP  SET
```

You select the program to be re-numbered using the CURSOR keys and alter the current number to the new number using the DATA control. You can also use the numeric keypad for this, setting the program number to a value of between 1 and 128 (these are MIDI specification limits). When you have altered the number, you have several options to choose from on soft keys 6, 7 and 8. These are:

ALL — This will set all programs to the same number as that set in the currently selected program. For example, placing the cursor on 3 BASS and setting that to 1 would re-number all the programs to program number 1. This is useful when you have loaded in a variety of sounds from different disks or hard disk volumes and wish to sequence them multi-timbrally (see later).

SLIP — This causes all subsequent programs to be re-numbered with an offset to the currently selected program. For example, if you re-numbered program 4 to 10, all subsequent programs (i.e. SYN BASS and others off the screen) will be re-numbered 11, 12, 13, etc.. whilst programs 1, 2 and 3 would be unaffected.

SET — This will only re-number the currently selected program. All other programs will be unaffected. You may also use this to re-number several programs in one go. For example, change the programs accordingly (i.e. 1, 2, 3, 4, 5, etc.) and then press SET

Once you are happy with the result of your re-numbering, press SLCT to return you to the main SELECT PROG screen display. Also, if you change your mind and do not wish to re-number something now, simply leave the RNUM page without pressing F6, F7 or F8.

MIXING SOUNDS ON THE S3200

The S3200 is equipped with a 'virtual mixer' - that is, you can set up mixes of programs with any number of mixer 'channels'. This is done in the MIX page of SELECT PROG. To access this, press MIX and you will receive the following display:

```
MIX    prog no: 1   loud st pan send lev
*  1 STRINGS 1        80 99 MID   1   45
   2 PIANO            76 99 MID   2   80
   3 BASS             99 99 MID   4   93
   4 BIG KIT          94 99 MID   3   89
   5 SYN BASS         87 95 MID   5   80
SLCT RNUM MIX MIDI DISK DEL RVB MUTE
```

On this page you may set, from left to right across the screen:

prog no: This shows the number of the program selected in the main SELECT PROG page although you may select another program or group of programs in this field.

loud: This sets the overall level for the program as it appears at the stereo output AND the individual output and could be regarded as similar to the gain control on a mixer. Please note, however, that if this parameter is set to 99, then you will lose control over velocity sensitivity of loudness. The default is set to 80 which gives a healthy output level and a good degree of velocity sensitivity.

st: This sets the level of the program as it appears at the left/right stereo outputs of the S3200. This would normally be used to mix the levels of different programs and is the equivalent of a mixers fader control. It is possible to send programs to individual outputs but, by mixing them out of the stereo outputs by setting this parameter to 00, you remove them from the main mix. In this way, for example, you could send individual drums to separate channels of an external mixer for more elaborate level and tonal control whilst other instruments appear only at the stereo outputs of the S3200. In this way, very complex mixes can be set up. Another method may be to send, say, snare, kick and hi-hats to individual outputs (but take them out of the stereo mix) and just have toms and percussion in a stereo image coming out of the main stereo mix. In this way, you save on channels on your external mixer as well as freeing up the S3200's other individual outs for maybe piano, bass or whatever other instruments you may have in a multi-timbral setup.

pan: This sets the pan position of the program in the stereo outputs and the range is L50 through MID (00) to R50.

send: This allows you to assign any program to one of the 8 individual outputs should you wish to mix the program using an external mixer.

> ***NOTE**: It is possible to send individual keygroups to these outputs so the results you get may sometimes be a little unpredictable if you are not sure of the keygroup assignments.*

You may also select REB or FX and you may use this to send programs in proportionate amounts to the internal effects and the lev: parameter explained next allows you to set the effects send level. You may also send the program to both effects sections in parallel by selecting R+F .

lev: This sets the level of the audio appearing at the individual outputs. If REB or FX or R+F is selected in the send: field described above, this control sets the effects send level to the internal effects.

All these parameters are available for each and every program and in a layered or multi-timbral setup, you can set levels and effects sends very precisely and the biggest difference between the S3200's mixer and a 'proper', external mixer is that the S3200's mixer does not have any EQ facilities for affecting tone. If you feel this is necessary, then you may use the individual outputs to send certain programs to an external mixer for more elaborate level and tone control and, of course, it is possible to use a combination of the external mixer and the internal mixer using the individual outputs in conjunction with the main stereo output.

MIDI

Pressing F4 - MIDI - will display the following screen:

```
MIDI prog no:  1 cha range pol pri  tr
*  1 STRINGS 1    1 C_0 G_8 32 NORM +00
   2 PIANO        1 C_0 G_8 32 NORM +00
   3 BASS         1 C_0 G_8 32 NORM +00
   4 BIG KIT      1 C_0 G_8 32 NORM +00
   5 SYN BASS     1 C_0 G_8 32 NORM +00
 SLCT RNUM MIX MIDI DISK DEL RVB MUTE
```

This could be regarded as a 'MIDI mixer' as it follows a similar layout to the MIX page described above. This page allows you to set various MIDI parameters for each program. The parameters are:

prog no: This shows the number of the program selected in the main SELECT PROG page although you may select another program or group of programs in this field.

cha: This allows you to set the MIDI channel for any program and the range is OM (omni) through 1-16. This parameter allows you to layer programs together by setting two or more programs to the same program number and setting their MIDI channels the same in this field. It also allows you to set up sophisticated multi-timbral configurations (i.e. set several programs to the same program number and assign different MIDI channels to each program). You may also layer two or more programs within a multi-timbral configuration of course.

range:

When two programs are given the same program number, it is possible to set up keyboard splits by setting the note range of the programs. This parameter ignores each programs keygroup ranges and simply imposes a range on the whole program. In this way, you can quickly set up very complex keyboard splits and layers without having to concern yourself with the intricacies of setting keygroups within a program itself. Please note that this parameter has no effect on single programs, it is only when two or more programs have the same program number that it is effective.

pol:

This is an abbreviation of POLYPHONY and allows you to limit the polyphony of a program. This is sometimes desirable in certain types of programs such as hi-hats, for example, where you want a closed hi-hat to shut off an open hi-hat. In this case, you would set the polyphony to 1. Similarly, you may wish a monophonic bass part to have a restricted polyphony.

pri:

This is an abbreviation of PRIORITY and allows you to set how notes will be 'stolen' by other programs if the 32 voice polyphony is exceeded. There are four settings: LOW, NORM, HIGH and HOLD. If the program is set to LOW priority, then notes from this program will be stolen first. If set to HIGH, then notes from other programs with lower priority will be stolen first before they are stolen from this program. NORM is, of course, normal priority and sets standard dynamic voice allocation and note stealing will take place with no particular priority. If a program's priority is set to HOLD, then notes can only be stolen from this program by the same program.

If you are playing a complex piece of music using many programs in a multi-timbral configuration, it is a good idea to set important programs to HIGH or HOLD and less important, background programs to LOW. If the piece of music is not overly complicated and polyphony is not going to be exceeded, you may prefer just to leave the priority at the default setting of NORM.

tr:

This is an abbreviation of TRANSPOSE and sets the basic octave range for the program. The range is +/- 50 semitones. You will note that this is not a pitch shift function as such but a MIDI transpose function - this overcomes the problem of playing back samples out of their range. What this function does is introduce an offset so that, even if you play C3 on the keyboard, this is offset to play the samples on C4 (with a +12 setting) - it is not playing the samples on C3 an octave higher.

DISK OPERATIONS

The next key along, F5, gives you access to some basic disk functions for loading sounds into the S3200. Pressing DISK in the SELECT PROG mode gives you this screen:

```
LOAD FROM DISK : FLOPPYH  vol: NOT NAMED
 STRINGS 1               |
 SOFT STRINGS            | programs:    8
 OCT STRINGS             | (samples:    6)
 SLOW STRINGS            | free mem: 100%
 PIZZA STRNGS            |          rLOADˀ
SLCT RNUM MIX MIDI DISK DEL P+S VOL
```

Here, you have a choice of two options: loading a particular program and its associated samples P+S or loading the entire contents of the disk VOL.

If you have inserted a disk, pressing DISK will bring up a list of all programs stored on that disk. If you have inserted the wrong disk or wish to change it, insert a new disk and press the DISK key again. If you want to wipe out all programs and samples in memory and load the contents of the disk, press VOL. You will be asked if this is what you really want to do. Make sure that you either do not want the programs and samples in memory, or that they are saved to disk before you proceed. As the disk is loading, you will receive this display

```
                        |STRING C3        S  6%
 loading sample:- STRING C2
```

indicating the loading progress.

The other option, P+S, allows you to load a program and all associated samples (free memory permitting). To do this, highlight a program using the CURSOR keys, and press P+S. The program, and its associated samples, will be loaded into memory. If the samples required by this program already exist in memory, they will be loaded anyway, but they will overwrite the samples currently in memory. If there is not enough memory to load a program and its samples, the message

```
                        |STRING C3        S  6%
 !! Insufficient waveform memory!!
```

will be displayed. In this case, you will have to delete some existing programs and/or samples to free up some memory space.

LOADING FROM HARD DISK

If you have a hard disk of any description connected via SCSI for loading sounds, you will receive the following display when F5 is pressed:

```
LOAD FROM DISK : HARD-:A  vol: NOT NAMED
 STRINGS 1               |
 SOFT STRINGS            | programs:   8
 OCT STRINGS             | (samples:   6)
 SLOW STRINGS            | free mem: 100%
 PIZZA STRNGS            |          ┌LOAD┐
SLCT RNUM MIX  MIDI DISK DEL  P+S  VOL
```

This is almost the same as for floppy except that you can see it tells you that it is a hard disk. You may select from different volumes by moving the cursor to where it says `NOT NAMED` and scrolling through the volumes on the disk. By moving the cursor to where it says 'A' after `HARD-:`, you may select different partitions to choose other volumes. Loading is done in the manner described above - press VOL to load the entire volume and P+S to load a particular program and its associated samples.

***NOTE**: There is more to using a hard disk than that, unfortunately, such as matching SCSI ID's, formatting, etc.. For details on using the S3200 with a hard disk, please refer to the section HARD DISK CONTROL in the DISK section.*

Other disk operations, like saving and formatting disks, are performed in the DISK mode, not from this page. This page is intended purely for quick access to loading from disk.

DELETING PROGRAMS

Programs and their associated samples may be deleted from memory in this page, which is accessed by pressing the DEL key. Pressing this key displays this screen:

```
DELETE PROGRAMS FROM MEMORY
*  1 STRINGS 1          | programs:   8
   2 PIANO              |
   3 BASS               |    free:  12%
   4 BIG KIT            |
   5 SYN BASS           |  ┌ delete ┐
SLCT RNUM MIX  MIDI DISK PROG PNUM ALL!
```

When this page is displayed, the cursor will highlight a program. Highlight the program you want to delete using the CURSOR keys. There are three soft key actions that you can take, all concerned with deleting programs: PROG PNUM ALL!

***NOTE**: If you delete a program from memory, make sure that you really do not need that program in the future, or that you have saved it to disk first. In some cases, as explained below, deleting a program will also delete samples. Make sure these really are unwanted or have been saved to disk before proceeding.*

The three 'action' keys are as follows:

PROG — Pressing this will display this prompt:

```
  5 SYN BASS          |    ┌ delete ┐
delete one program?             GO ABORT
```

and you should press GO or ABORT as necessary.

If the program is the only one in memory using a particular set of samples, you will receive the following prompt:

```
  5 SYN BASS          |    ┌ delete ┐
delete   3 released samples?   NO   YES
```

Press the appropriate soft key if you are sure that you want to delete the program and its associated samples.

If the samples contained in the program are used elsewhere in other programs, you will not receive this prompt as it is assumed you don't want to lose these samples.

PNUM — This will delete all programs which have the same number as the highlighted program. You will be asked if you want to proceed with the bulk program delete. If these programs are the only ones using particular samples, you will be asked if you want to delete the samples as well.

ALL! — This, of course, is the most drastic of these three options. If you answer GO and YES to the questions regarding released samples, then all programs, (except for an S3200 generated program - TEST PROGRAM) and samples will be deleted. Deleting samples and rearranging memory space will take a little time, so be patient while this takes place.

***NOTE**: Obviously, this is an option to be used with some caution. If the programs and samples in memory have not been saved to disk, deleting programs and/or samples at this point will be fatal - YOU WILL NOT BE ABLE TO RETRIEVE THEM. Please be careful.*

USING THE S3200'S EFFECTS

These days, effects are as important a feature of a sound as the filter settings or envelope settings and when you load a sound from disk you want to be able to recall it with its effects every time you load it. The S3200's two internal effects sections allow you to do this.

When you turn on the S3200, you have two effects sections RVB and FX. RVB is an abbreviation of REVERB and gives you access to 50 reverb effects and the FX selection gives you access to 50 effects comprising multi-tap echo, stereo chorus/flanging, stereo pitch shift and a single delay line with modulation. These may be used in parallel so that, in a multi-timbral or layered setup, some programs may be send to one section whilst other may be sent to the other. It is also possible to send one program to both effects sections simultaneously. These 100 preset effects are made up of five basic effects types. These are:

REVERB - This allows you to select from several different types of reverb from long, spacious halls to tight rooms.

ECHO - this is a three tap delay line. In other words, instead of having just one delay setting as most units do, you have three and each delay can be set separately each with its own feedback and pan position. This allows you to create a wide range of delay and echo effects from straight single delays to ping-pong echo through to complex multi-tap echoes that can simulate the echo effects only offered by older tape echo devices but without the wow and flutter and tape hiss, of course!!

CHORUS - there is one all purpose effect that covers every type of modulated delay effect from mild, shimmering chorus to flanging effects. The algorithm used for these effects is very complex and uses four delay lines that are modulated by a low frequency oscillator (LFO) but the modulation phase angle for each delay line is different. This allows you to create rich, swirling stereo effects and also eliminates the unpleasant cyclic repetition you get with chorus and flange units that use only one delay with one LFO.

PITCH SHIFT - this is a stereo pitch shifter that allows you to transpose a sound up or down by as little as .01 of a semitone for subtle detune effects to 50 semitones. There are two pitch shifters and each one has a delay line in the feedback loop allowing many interesting special 'spiral' and arpeggio effects to be created.

DELAY - This emulates a single delay digital delay line for long delay effects. Modulation is also provided for flanging effects.

On the S3200, you have two 'EFFECTS FILES' and each contains 50 effects. The effects in each of these effects files can be freely assigned to any program number so that one effect can be used on several different programs. For example, you may have a strings program, a brass program and a piano program with the program numbers 1, 2 and 3 and you may wish them all to share the same chorus effect. In this case, all you need to do is assign the appropriate chorus effect to programs 1, 2 and 3. Of course, every program can have its own unique effect if you wish.

Similarly, a group of programs that have the same program number may share one effect so that, in a layered, split or multi-timbral setup, you may assign an effect to all sounds 'globally' and each program can have its own effects level using the SEND parameter described in the mixer section above.

Furthermore, because there are two separate effects sections, different programs may be sent to different effects. One program may be sent to two effects simultaneously or, in a layered or multi-timbral setup where several programs share the same program number, different programs may be sent to different effects. In such circumstances, it is also possible for some programs to go to one set of effects, other programs to go to the other set of effects and some programs to go to both. This allows a great deal of flexibility.

This method of effects assignment makes the internal effects unit behave more like an external unit where effects can be freely assigned to any program and you can mix and match your effects to programs as you like. If you don't like any effect, simply select another until you find one that matches the sound exactly (or, of course, create your own). Furthermore, it is possible to 'grab' an effect off another disk and assign it to any program.

> ***NOTE**: When you use the following internal effects: reverb, chorus, pitch shifter and delay, the polyphony is:*
>
> *With reverb: 30 voices*
> *With chorus and pitchshifter: 28 voices*
> *With delay: 31 voices*

To access the multi-effects, press F7 - REB - and you will receive something like the following screen display:

```
REB(prog:  1= 1) no: 1 LONG HALL 1
     type: LARGE HALL           output: 99
    decay: 37                      pan: MID
  HF damp: 98                   HF cut: 99
    delay: 48mS                  width: 99
  diffuse: 99
SLCT RNUM  MIX  MIDI SAVE COPY  FX  MUTE
```

The fields across the top of the screen are as follows:

`prog:` This field allows you to assign any effect to any program. This is done by selecting the appropriate program (this would normally be done in the main SELECT PROG screen but can be changed here if you wish by changing the first numeric field) and then assigning the effect you require. In other words, if the display reads:

```
(prog:   3= 4)
```

then program number 3 has effect number 4 assigned to it. You can change the effect assignment by changing the effect number and, as you do this, the name of the effect shown alongside will also change.

`no:` This shows the currently selected effect number. You can change this to audition other effects without necessarily changing the effects assignment set in the `prog:` field described above. If you prefer the effect assigned in this field, you can then assign that effect in the `prog:` field. This field also allows you to temporarily assign another effect but this will not be retained when you select another program.

effect name — Although not labelled as such, this field shows the name of the selected effect. You may create your own name by pressing NAME and typing in a name from the front panel and then pressing ENT/PLAY. Names of up to 12 characters can be used. As you select different effects in the prog: or no: field, the name shown here will change.

The rest of the screen depends on the effect chosen in the type: field described in a moment but our designers have kept many parameters consistent between different effects to make programming that much easier for you.

The parameters on the right of the screen are more or less constant for every effect.

output: — This sets the output level of the effect.

pan: — this sets the pan position of the effect.

HF cut: — this is a pre-EQ that limits the amount of high frequency component going to the effect. This allows you to filter out high frequencies and so create smoother effects. (This does not appear in the ECHO effect)

width: — this allows you to set the stereo 'spread' of the effect. All effects are stereo and give wide, spacious effects but there may be occasions where you don't want that width. This control allows you to limit this. 99 represents full stereo width and 00 gives you a totally monaural effect and you may set a value anywhere in between to obtain the right effect (this does not appear in the DELAY effect).

These parameters are consistent between REB and FX and apply to each section independently. Each effect itself has different parameters which are described below.

REVERB EFFECTS SECTION

The reverb effects are the first one you encounter when you press the REB key (F7) and you will receive the following screen display.

```
REB(prog: 1= 1) no: 1 LONG HALL 1
    type: LARGE HALL          output: 99
   decay: 37                     pan: MID
 HF damp: 98                  HF cut: 99
   delay: 48mS                 width: 99
 diffuse: 99
SLCT RNUM MIX MIDI SAVE COPY  FX  MUTE
```

The parameters are:

type: — This allows you to select the type of effect and you have a choice of the various reverb types. There are 6 reverb types available, all of which offer different densities and decay times. The reverb types available are:

LARGE HALL — This is a big, spacious reverb effect that emulates the characteristics of a large concert hall or cathedral environment. It is well suited to strings and other orchestral instruments, choir, church organ or any other instrument that needs to 'placed' in a big acoustic environment.

MEDIUM HALL — This emulates a slightly smaller acoustic environment such as a concert hall although this has a slightly less diffused reflective quality than the large hall selection and is slightly brighter. Again, this is ideal for strings, orchestral instruments, choir, etc.. It is also effective for creating huge, ambient drum sounds.

LARGE ROOM — This has the characteristics of a big, reflective room. It's decay characteristics are such that discrete echoes can be heard slightly. This is ideal for drums and percussion, guitars, piano or, in fact, any sound that needs to be placed in a more 'intimate' acoustic environment.

SMALL ROOM — This has highly coloured reflective qualities such as you experience in a small, reflective room. This reverb type is ideal for adding 'space' but without the 'wash' of a longer reverb type and so is effective on drums and percussion, bass, guitar and keyboard sounds. The room size is probably too small to fit an orchestra in but try it if you wish!!

PLATE 1 — This is a highly reflective, bright metal plate reverb simulation and is well suited to drums and percussion although most instruments will benefit from this effect.

PLATE 2 — This is a richer plate reverb sound with less high frequency content. This is a smoother sound and, again, is well suited to a variety of instruments.

There are no hard and fast rules as to the type of reverb you choose for any one sound. It could be that one type of reverb that sounds great on one sound won't necessarily sound effective on another so please experiment. The parameters that effect these reverb types are:

decay: — This sets the decay time for the reverb effect and the range is 00-99 although, within each reverb type, certain limits have been imposed so that, for example, you cannot obtain a 20 second decay from a small room reverb type. The range of decay times is, therefore, appropriate to the type of reverb selected above.

HF damp: — This sets the amount by which high frequency signals will decay proportionate to the main decay. In natural reverb, it is common that high frequencies will decay before lower frequencies because high frequencies have less energy than low ones and so are absorbed more quickly in the reflective process. This control allows you set this decay rate (or 'absorption rate', if you like) of the high frequencies to suit the type of acoustic

environment you you are trying to emulate. The rule of thumb is that for reflective environments set a low HF damp factor and for less reflective reverb types (i.e. for an acoustic environment that maybe has many 'soft' fixtures such as drapes, curtains, etc.) set a higher HF damping factor.

delay: This allows you to set the delay between the direct signal and the onset of the reverb decay. In very large acoustic environments, it is common that there will be a delay between the direct sound and the onset of the reverberation. This is known as 'pre-delay' and is set by this parameter.

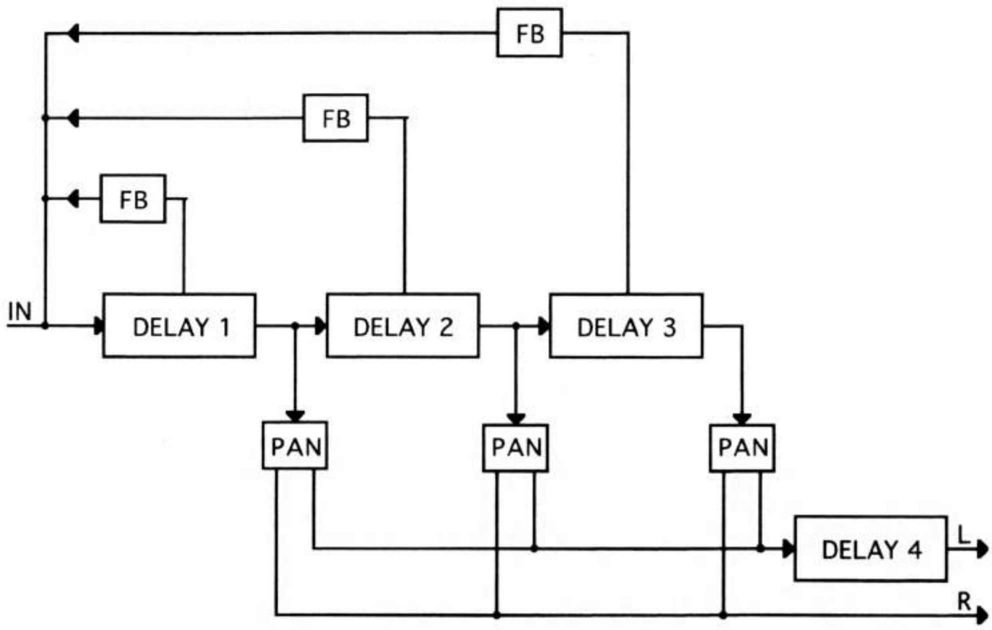
diffuse: This parameter allows you to set the 'smudge factor' of a reverb effect. Some acoustic environments are extremely reflective and you may hear discrete echoes. Other environments cause the reflections to be more erratic and so smooth out discrete echoes. The diffuse parameter allows you to set how reflective or smooth the reverb pattern is during its decay. Low settings of diffuse create a more reflective environment where echoes can be heard and higher settings smooth out these reflections into more of an ambient 'wash'.

EFFECTS SECTION

The second group of effects are accessed in the reverb page by pressing F7 - FX - and this will toggle between REB and FX accordingly. The first effect available in FX is ECHO.

ECHO

The echo effect allows a great deal of flexibility in the creation of multi-tap delays.



There are four delay lines that feed each other in series. Delays 1 - 3 are capable of up 360mS of delay whilst delay 4 is capable of 180mS of delay and, in total, you have 1260mS of delay. Each delay has its own feedback loop and pan position control and repeat echoes have the effect of 'flying' round the stereo image. These parameters allow you to create a wide range of echo effects that emulate the warmth and density of tape delay units with their complex, multi-head echoes.

To further assist in the simulation of natural echo and tape echo, there is a high frequency damping parameter that gradually reduces high frequency components so that multiple repeats get slightly duller with each repeat. There is also a delay on the left output.

When you select ECHO in the type: field, you get the following display:

```
FX (prog: 1= 1) no: 1 PING PONG
     type: ECHO                   output: 99
      D1 > D2 > D3 > Dleft           pan: MID
del:  360  360  360      0 mS
fbk:   00   00   50 damp: 00       width: 99
pan:  R50  R50  L50
SLCT RNUM MIX MIDI SAVE COPY RVB MUTE
```

and the parameters are as follows:

`del:` This sets the delay time for each of the delay lines 1 to 4. The range for delays 1 to 3 is 0-360mS each and for delay 4, the range is 0-180mS.

`fbk:` This sets the amount of feedback for each delay. Please note that at excessive settings, the feedback has the potential to overload and cause instability.

`pan:` This sets the pan position for each of the delay lines.

`damp:` This sets the high frequency damping so that each repeat gets progressively duller. This is a phenomenon that happens with natural echo and can be easily simulated here.

`output:`, `pan:` and `width:` are the same as for the other effects.

CHORUS EFFECTS

These are modulated delay line effects otherwise known as chorus and flanging. The principle behind such effects is that a short delay line is modulated by a low frequency oscillator (LFO) and this has the effect of creating pitch and tonal variations that can be used to add richness and depth to a sound. As mentioned, the algorithm for these effects uses four delay lines each with an out of phase LFO for modulation resulting in a rich, non-cyclic effect. Classic flanging effects are available using the DDL effects explained later. The chorus effects are selected in the `type:` field. Selecting this effect gives you the following screen:

```
FX (prog: 1= 1) no: 1 CHORUS
   type: CHORUS                output: 99
                                  pan: MID
  speed: 10                    HF cut: 99
  depth: 50                     width: 99
feedback: 50
SLCT RNUM MIX MIDI SAVE COPY RVB MUTE
```

The parameters are as follows:

`speed:` This sets the LFO rate for the effect. The LFO rate can be set between 01 and 99 and at 00 it is off. When switched off, the `depth:` control, in conjunction with the `feedback:` control, allows you to set the chorus or flange effect manually and you can use this to add interesting, non-harmonic metallic overtones to sounds.

`depth:` This sets the depth of modulation from the LFO and the range is from 00 to 99. If the LFO is set to 00 (i.e. off), this control allows you to 'tune' the metallic overtones to specific pitches. When `depth:` is set to 00, no chorus or flange effect will be heard. When the `speed:` control is set to 00, you may use this control to set the non-harmonic overtones to the sound.

`feedback:` This sets the amount of output signal that is fed back into the input stages of the chorus. Increasing this creates a more dramatic effect. It is most useful in

flanging effects although small amounts of it will accentuate certain chorus effects. Be careful when using this control as it is possible on certain sounds to introduce harmonic instability - in other words, it may accentuate a certain frequency in the sound and cause loud peaks. In certain circumstances, these peaks will be re-circulated and may create undesired 'howl round'.

PITCH SHIFTER

The S3200 contains a stereo pitch shifter and it is possible to set separate pitch shifts for the left and right outputs. Furthermore, you may set delays within the pitch shifters feedback loop to create a variety of interesting arpeggio effects.

Selecting PITCH SHIFT in the TYPE field calls up the following screen:

```
FX (prog:  1= 1) no: 1 MILD DETUNE
     type: PITCH SHIFT          output: 99
           LEFT    RIGHT           pan: MID
     tune:-00.05  +00.05        HF cut: 99
 feedback:  00      00           width: 99
    delay:   0 mS    0
 SLCT RNUM  MIX  MIDI SAVE COPY  RVB  MUTE
```

As you can see, there are separate controls for the left and right pitch shift. These are:

`tune:` This sets the pitch shift and is variable between 00.01 of a semitone to 50 semitones up or down.

`feedback:` This control sets the amount of signal that is re-circulated back into the pitch shift. Be careful because with certain sounds and certain pitch shifts you may get instability and 'howl around'.

`delay:` This control sets a delay time for the feedback parameter. At higher settings, the signal feeding back will be delayed and so, using this parameter, it is possible to create a wide range of arpeggio and other effects. With the `tune:` parameters set to a wide interval and feedback set to, say, 60, you can create rising and falling arpeggios. With the tune controls set to a smaller interval, you can create interesting echo effects where there is a slight pitch bend on the repeats.

These parameters are identical between the left and right pitch shifters although they are totally separate. The remaining controls are the same as the other effects.

DELAY

This is a standard DDL (digital delay line) that offers one single repeat. Selecting DELAY in the type: field will display this screen:

```
FX (prog: 1= 1) no: 1 DDL
    type: ECHO                  output: 99
   delay: 500 mS                   pan: MID
 feedback: 50                   HF cut: 99
 lfo rate: 10
lfo depth:    0 mS
SLCT RNUM MIX MIDI SAVE COPY RVB MUTE
```

This effect type allows you to set up long, repeating delays of up to 1 second as well as thick flanging effects. The parameters are:

delay: This sets the initial delay time and is variable from 0-999 milliseconds.

feedback: This sets the feedback for the delay. When used as a DDL for echo effects, this sets the number of repeats. When used with LFO modulation for flanging effects, this emphasises the effect. Be careful with this control - as on all DDL's, setting it too high may cause instability!

lfo rate: This sets the modulation speed for flanging effects.

lfo depth: This sets the depth of the LFO when creating flanging effects. It is measured in milliseconds - i.e., it shows how many milliseconds the LFO will sweep through.

To set delays, simply set the delay: parameter and the feedback: parameter. To create flanging effects, the delay time really needs to be set between 5 and 20 milliseconds. The lfo rate: and lfo depth: can be set to anything and the feedback: parameter can be used to to emphasise the effect making it thicker and nastier.

MUTING EFFECTS

There is one final function that applies to all effects and that is MUTE. This is available in all of the SELECT PROG modes and allows you to temporarily disable the effects so that you may hear your sound 'dry'. This parameter is not stored as part of the program and is a local function only although its status is saved to disk as a volume attribute.

Normally, this soft key will display MUTE. To mute the effects, press F8 and you will receive the display >><< in this 'soft box' and the effects will be muted. Pressing it again will restore the MUTE message and the effects will be switched back on.

COPYING AND MOVING EFFECTS AROUND

In the FX mode, it is possible to move one effect to another location using the SAVE and COPY soft keys (F5 and F6). Select the effect you wish to move and press SAVE (F5). The effect is placed in a small 'clipboard' and the cursor will automatically be placed on the No: field and you may now select the effect number in which you wish to place this effect. When you change this number, the effects selection changes to show you which effect you will be

overwriting. When you have found the required location, press SAVE - F6 - and the effect will be copied.

You may also take effects from another disk using this function. Ensuring that you save your current effects file first, load the required effects file from a disk. When the effects file has loaded (it will only take a second or two), select the the effect you wish to use and save it. Now remove the disk and load up the original effects file and choose a location and copy it. In this way, effects from any disk can be loaded into the S3200 and used with any program you wish.

ASSIGNING MULTIPLE EFFECTS TO PROGRAMS

Because the S3200 has two separate effects sections, it is possible to assign programs to different effects or to assign one program to two effects in parallel. There are limitations to this however, and you may only use one or the other group of effects. For example, you may freely assign programs to reverb and chorus, reverb and delay, reverb and pitch shift but you cannot assign programs to delay and chorus, pitch shift and delay, etc.. You may only combine effects from the two separate sections. Neither is possible to combine effects in series (i.e. delay INTO reverb is not possible).

To assign, say, an electric piano sound to chorus and drums to reverb, select one of the chorus effects and assign it to the piano program and select a reverb effect and assign it to the drums program(s). Now, in the MIX page (or in the OUT page of EDIT PROGRAM (see later)), set the send parameter for the piano to FX and the same parameter for the drums to REB. The lev parameter will dictate the level going to each effect section.

To assign both effects sections to one sound (for example to apply reverb and delay to a guitar sound) select appropriate effects in either effects section and select F+S in the MIX page (or the OUT page in EDIT PROGRAM). Now the sound will be sent to both effects at a level set by the lev parameter. To adjust the balance of each effect, use the output: parameter of each effects section.

USING SELECT PROG - PROGRAMS WITH THE SAME NUMBER

Most of the time, you will probably want to load in the sounds you need and these will be single programs for playing from the keyboard (or whichever MIDI instrument you play). As we have seen, this is easy enough - simply scroll through using the DATA control or select them via MIDI program change. There are occasions when you want to play several programs together, however, and this is where the re-numbering becomes useful.

LAYERING PROGRAMS

The S3200's 32-voice polyphony makes layering a more viable proposition than on 24- or 16-voice samplers. Although it is possible to layer sounds within one program (something which may be more convenient on occasions and one we will look at when we venture into EDIT PROGRAM), it is useful to be able to call up several programs together.

For example, let's say you have a string program and brass program you want to combine. As mentioned, you could go to great lengths to make up a program that combined all of these samples but an easier way is to simply give them the same program number - for example, re-number them both to, say, program 1. When you select program 1 you will then select the combined programs with the strings and brass layered together. You may also set the balance and pan positions between these sounds in the MIX page.

You may prefer to layer sounds using RNUM rather than creating one program with the required sample(s) in it because there are certain aspects to creating a program that affect all things the same. For example, the auto-panning and effects send would affect all samples equally - when using the RNUM function, one sound in the layer could be panning slowly left to right with lots of echo whilst another stays central with no effects. You also have easier control of the respective levels of the programs in the layers.

Of course, you are not limited to layering just two sounds - you could layer up to 32 programs for a monster monophonic lead line or bass sound!

CREATING KEYBOARD SPLITS - METHOD 1

As for layering, it is easy to set key splits in EDIT PROGRAM but, again, it may be more convenient to create the split using the RNUM function.

Let us say you have two programs - UPRIGHT BASS and VIBES - and you want to create a split so that you can perform a walking bass line with your left hand and a jazzy vibes solo with the right. By setting the RANGE parameter in the MIDI page (F4) accordingly, the bass can be set to finish at B2 and the vibes set to start at C3. Even though both these programs originally spanned the entire keyboard, this function allows you to impose limits on the programs' keyranges. Now, assuming that both programs have the same program number (i.e. both are re-numbered 1 or something), then you can play both programs with the keyboard split.

CREATING KEYBOARD SPLITS - METHOD 2

This second method requires you to select two programs whose keyranges were limited in EDIT PROGRAM when the programs were created. Let us say you have a bass guitar program that only spans as far as C3 and a solo flute whose keyrange starts at C3 - simply re-numbering those programs so that they share the same number will create the keyboard split (although in this example, playing C3 will play both programs).

USING RENUMBERING TO CREATE MULTI-TIMBRAL SETUPS

One of the most appealing things about MIDI is its multi-channel ability. Originally, synth modules could be set to a specific MIDI channel number so that several modules could be set to play several musical parts from a sequencer. Of course, as technology advanced, it became possible to do this within one module and such a module is known as 'multi-timbral' - i.e. 'many sounds'. The S3200 is no exception and has powerful multi-timbral capabilities.

To set up a multi-timbral assignment, we use the re-numbering function again. Assuming you have loaded in a pile of programs you wish to sequence, give them all the same program number. This is done in the RNUM page and you simply give the same number to all programs using the ALL function. You should receive a display something like this when you return to the main SELECT PROG screen:

```
PROGRAMS IN MEMORY (vol: VOLUME 020    )
*  1 STRINGS 1      |   8 program(s)
*  1 PIANO          |   8 now active
*  1 BASS           |  PROGRAM NUMBER:  1
*  1 BIG KIT        |
*  1 SYN BASS       |
 SLCT RNUM MIX  MIDI DISK DEL  REB  MUTE
```

Here, all programs are number 1.

Of course, you will need to assign each program to have different MIDI channels:

```
MIDI  prog no:  1  cha  range  pol pri   tr
*  1 STRINGS 1      1  C_0 G_8  32 NORM +00
*  1 PIANO          2  C_0 G_8  32 NORM +00
*  1 BASS           5  C_0 G_8  32 NORM +00
*  1 BIG KIT       10  C_0 G_8  32 NORM +00
*  1 SYN BASS       7  C_0 G_8  32 NORM +00
SLCT RNUM MIX  MIDI DISK DEL  REB  MUTE
```

and you can use the MIX page to set the levels, pan position, output assignment and effects send for each program:

```
MIX    prog no:  1   loud st  pan send  lev
*  1 STRINGS 1         80  99 MID   1    45
*  1 PIANO             76  99 MID   2    80
*  1 BASS              99  99 MID   4    93
*  1 BIG KIT           94  99 MID   3    89
*  1 SYN BASS          87  95 MID   5    80
SLCT RNUM MIX  MIDI DISK DEL  REB  MUTE
```

Remember, also, that you may use a combination of all of these techniques and you can, of course, have a mix of split and layered programs within a multi-timbral assignment.

Using the S3200 live, you may have several multi-timbral setups in memory at any time (memory allowing, of course!). In this way, you may have sets of programs for each song recalled from a MIDI program change from your sequencer.

And don't forget that these programs may utilise the internal effects sections for that final finishing touch.

Once you are happy with anything you set up, remember to save it to disk if you want to keep these changes.

If you are new to the S3200, don't worry about all this for the moment - as you gain more experience with the instrument, come back to these techniques when you're ready. They will also make more sense in a moment when we have a look at some of the other functions.

CONCLUSION

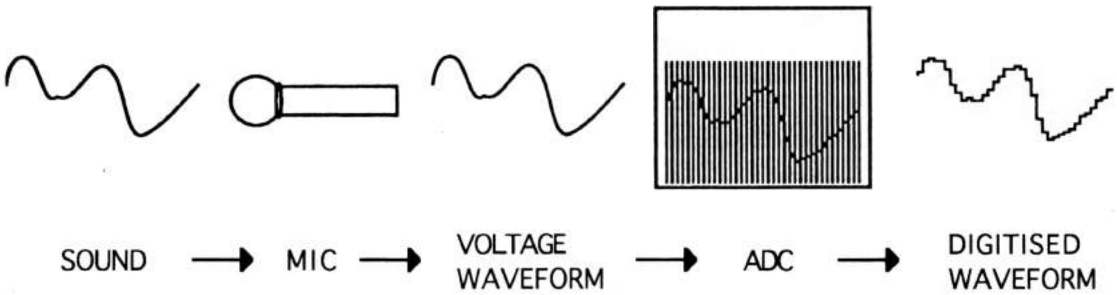
So far, we have seen the enormous possibilities offered simply by loading some sounds off disk, re-numbering them, mixing them and playing around with the effects. There are even more exciting possibilities open to the adventurous musician when we enter the realms of sampling and creating programs and this is the real joy of owning a sampler. The next section covers all this so, if you're ready.....

EDIT SAMPLE - RECORDING AND EDITING

Edit sample is where you make and edit your recordings. In this mode, you may, amongst other things, trim, loop, join, merge, reverse, re-sample and timestretch recordings. But before we can do any of that, we first need to make some recordings.

WHAT IS SAMPLING?

Sampling is a process where we record sound digitally. All natural sound comes in the form of variations in sound pressure. Using a microphone, we can convert those changes in air pressure into rising and falling voltages. Once they exist in that format, we can process them through ANALOGUE TO DIGITAL CONVERTERS (ADC) to turn those voltages into streams of digital data. Once they exist as digital data, we can edit them with alarming precision.



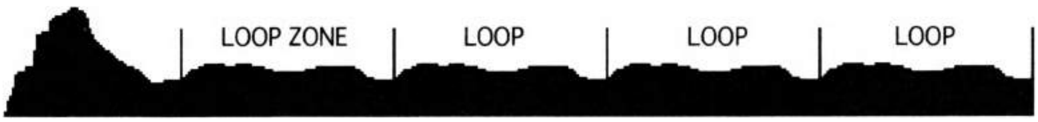
In the ADC, the sound is 'sampled' at a rate of 44,100 times per second. You can liken this to film. If we take a lot of photographs in very quick succession, when we play it back, we have the illusion of movement. The same is true of sampling. If we take enough samples, we get an accurate reproduction of the sound. To carry this analogy even further, if you think of the very early days of film where they didn't take so many frames in a second, the results were jerky and distorted. The same could be said about old samplers - because they sampled less (that is, the sampling rate was lower), the sound quality was not so good. In order to reproduce sound accurately, you need to sample at a frequency that is at least twice the upper reaches of the sounds frequency range. In other words, if a sound contains frequencies that extend to, say, 15kHz, you need to sample at 30kHz at least. Instruments such as cymbals which are very bright and contain many overtones need to be sampled at 40kHz. A bass drum, however, which has very few upper harmonics, could feasibly be sampled at 20kHz.

The S3200 samples at 44.1kHz, the same as compact disc so you can be sure you are getting CD quality sound from your sampler.

The digitized waveform is loaded into RANDOM ACCESS MEMORY (RAM) where it exists as numbers. As you know, computers and microprocessors are very good at dealing with numbers and so we are able to rearrange those numbers and so alter the sound.

At the end of the process, we need to be able to convert those numbers back into an electrical analogue waveform and so the numbers are reconstituted into analogue via DIGITAL TO ANALOGUE CONVERTERS and output to your mixer or amplifier.

One of the inherent problems with sampling is the RAM and it is not possible to have an endless supply of it installed in the sampler. As a result, our recordings (or 'samples' as they are more commonly known in the music industry) have to be kept fairly short. In order to make them last longer so that long notes can be sustained, we need to loop them. This involves selecting a portion of the sound that will repeat over and over again when we hold our finger(s) on the keyboard.



The biggest problem, though, is the nature of the sounds you sample. Most musical instruments have particular resonant frequencies (or 'formants' as they are sometimes called) plus other characteristics such as vibrato, etc.. In the instrument themselves, these characteristics stay constant regardless of the notes being played but, on a sampler, because you are transposing the sound up and down (slowing it down and speeding it up much like a tape recorder), these are also transposed which leads to a phenomena quaintly referred to as "munchkinisation". We have all laughed at hearing our voice speeded up on a tape recorder sounding like some bizarre cartoon character - the same will happen on a sampler and this is because the voice has formants and other attributes which do not transpose well. Similarly, the sound's envelope will change - transposed down an octave, a percussive attack will sound quite sluggish.

To overcome this, we need to use a technique known as MULTI-SAMPLING - that is, taking various samples of the instrument at a variety of pitches across its range so that, at any one time, the sound is never transposed too much and so avoids serious "munchkinisation" and envelope distortion. Typically, you can get away with one sample per octave but some difficult instruments with strong formants need more. The saxophone and piano are two instruments that spring to mind and which are notoriously difficult to capture.

Another property of an acoustic instrument is that it can make so many sounds depending on how it's played. When played softly, the sound is not only quieter but softer in tone and, when played hard, is louder and brighter. Some instruments have quite extreme ranges in tone. Coupled with playing techniques (i.e. thumbed and slapped bass, bowed and plucked violin), to accurately replicate this on a sampler, we can take different samples according to playing styles. On the S3200 we have four velocity zones that allow us to use playing technique to switch between these different samples so that you could, for example, use velocity to switch between a slow legato viola and an aggressively bowed viola.

Of course, a lot of the time you can take just a few samples, loop them for sustain, map them out across the keyboard and have perfectly acceptable results which are usable in a wide range of applications. For some sounds such as drums and percussion, you don't need to loop.

Let's now have a look at the EDIT SAMPLE functions

THE MAIN SAMPLE SELECT PAGE

The page you receive on first entering this mode looks something like this:

```
SAMPLE EDIT              sample: PULSE
name: PULSE                size:       256
     *existing Samp*       Free: 2370832=100%
                          samples in mem:  6
mode: MONO               monitoring program:-
                              MONITOR
 SLCT REC1 REC2 ED.1 ED.2 ED.3          DEL
```

Here we see one of the default sample waveforms, a pulse wave called, not surprisingly PULSE. Also on this screen is information regarding the current state of your memory and you can see the size of the selected sample, how much memory is free and how many samples are currently in memory.

In EDIT SAMPLE, there is an alternative, temporary style of display that allows you to see sample length, etc., in milliseconds and you may toggle between the two types of display by pressing and holding down any of the page keys again. You may also press the page key again to toggle between displaying MIDI notes as numbers or as names. In this page, pressing SLCT will switch between sample point and millisecond displays and will display this screen:

```
SAMPLE EDIT              sample: PULSE
name: PULSE                size:       5MS
     *existing Samp*       Free: 39760MS=100%
                          samples in mem:  6
mode: MONO               monitoring program:-
                              MONITOR
 SLCT REC1 REC2 ED.1 ED.2 ED.3          DEL
```

NOTE: *The millisecond display is purely for reference - you cannot actually edit in milliseconds. This is not a fault of the S3200. The reason is that all editing has to be done referenced to sample points as this is the only true reference the S3200 has when dealing with samples. Because a sample can be played anywhere on the keyboard, to edit in milliseconds makes no sense - what does 5 milliseconds mean to a sample recorded at C3 but being played at C2? You will find this display option useful though when setting certain parameters as we shall see.*

Only two fields are available in the SLCT page: monitoring program: allows you to monitor a sample you are making or have made or are editing referenced to the program it is placed in. For example, let us say you have made a snare drum sample and this is placed into a program DRUMS 1. By selecting DRUMS 1 in this field, you will be able to edit it in EDIT SAMPLE whilst listening to the other drums in the program. For example, to trim the start of the snare so that it 'feels' just right, you may want to monitor the other drums as well, maybe even sequence them as you are editing. This field also allows you to play samples in EDIT SAMPLE on a MIDI channel other than 1, the default. The other field, mode:, allows you to select whether your editing and sampling will be in MONO or STEREO. This may be set here and in other pages if you wish.

The soft keys along the bottom are:

SLCT	This indicates you are on the sample select page. Press this to switch between sample points and millisecond displays.
REC1	This takes you to the record setup page where you may set such parameters as sample rate, sampling time, etc..
REC2	This takes you directly to the recording page.
ED.1	This takes to one level of editing giving you TRIM, LOOP and JOIN samples.
ED.2	This takes to another level of editing where you may timestretch and re-sample your recordings.
ED.3	This takes you to a third level of editing where you may perform 'sectional' editing (that is remove sections from samples), scale and normalise levels and set digital fades on samples.
DEL	This allows you to delete samples from memory.

These will be discussed in a moment.

NAMING SAMPLES - COPYING AND RENAMING

Before you can do any sampling, you must name the sample you are about to record. If you have just switched the S3200 on with no disk in it, you will have the four synth waveforms in there. Use one of these as the basis of your new sample.

To copy or rename a sample, press the NAME key - this turns the front panel keys into letter entry keys and you may type in a name of up to 12 characters (upper case only). The +/< and -/> keys input backspace and spaces accordingly and the MARK and JUMP keys input '#' and '.' respectively.

You will see this prompt:

```
LETTERS . . (NAME for numbers ENT to exit)
```

Pressing the NAME key again switches the numeric keypad from letters to numbers and you will receive this prompt:

```
LETTERS . . (NAME for letters ENT to exit)
```

You may press NAME again to access the numeric keypad's letters. When in the 'numbers' mode, the +/< and -/> keys input '+' and '-' to a name. Pressing NAME again reverts you to entering letters from the numeric keypad.

Alternatively, in conjunction with the CURSOR keys which can be used to move the cursor around within the name, you can use the DATA control to scroll through characters.

When you have entered your name, press ENT and you will get this prompt:

```
Select: COPY REN exit
```

Pressing COPY will copy the original sample - use this to create a basis for a new sample.

If the sample name is an existing one, the boxed area to the top left of the screen will show something like:

```
name: STRING C3
    *existing Samp*
```

and you will receive the following prompt:

```
!! MUST USE A DIFFERENT NAME !!
```

You must enter a unique new name.

Pressing REN will simply rename the currently selected sample with the name just entered. If the name exists, you will be prompted as above and you must re-enter a unique name.

Pressing exit will exit the naming process altogether with no action taking place.

It is worth taking the time to name your samples sensibly. It may be quick to simply call them SAMPLE 1, SAMPLE 2, 3, 4, etc., but, when you come to put them into a program, these names may not mean much to you. If you return to them in a week they certainly won't. Names like PIANO C#3 are best - this tells you the instrument and the note it is sampled on so it will be easier to set your program up later when you come to assign your samples to specific keyranges.

DELETING SAMPLES

It is possible to delete samples using the DEL key - F8. Pressing this will give you the following prompt:

```
delete one sample ?                GO ABORT
```

and you should press F7 or F8 accordingly.

***NOTE**: Deleting samples is destructive. Please ensure that you have saved them to disk before deleting in case you want to come back to them at a later date.*

SETTING UP FOR A RECORDING

There are two pages, REC1 and REC2, which deal with making recordings. REC1 is a 'record setup' page where certain parameters may be set if necessary. REC2 is where you actually make your recordings although certain important parameters from REC1 are also available there so you may prefer to go straight to REC2 to make a recording.

Pressing REC1 will display this screen:

```
RECORD SET-UP sample name: STRING C4
   mode: MONO        |    *existing Samp*
 (V)iew: LEFT        | bandwidth: 20kHZ
  start: INPUT LEVEL |orig.pitch: C_4
monitor: ON          |record tim:   1.00s
 (F)ree: 2257360=100% | =   44100=    1%
SLCT REC1 REC2 ED.1 ED.2 ED.3        DIGI
```

This shows that you have named (or selected for over-writing) a sample called STRING C4. You may select another sample for recording over if you wish (assuming they exist, of course!) by moving the cursor to the sample name field and scrolling through the available samples.

The fields on this page are:

`mode:` Here you may select between stereo or mono recording and editing. If you select STEREO, the sample will automatically have -L and -R appended to both the left and right samples respectively after the recording has been made. After that, any editing you do will be done in stereo unless you specifically switch to mono.

NOTE: *ED.2, editing is only in MONO. ED.1 and ED.3 can use stereo editing.*

`(V)iew:` Here you may select which side of the stereo image you wish to look at when editing stereo samples. You will note that if MONO is selected above, you cannot select anything other than LEFT. The 'V' is in parentheses because this field is shown in other pages of the EDIT SAMPLE mode as an abbreviation.

`start:` This field allows you to select how recording will be initiated. The choices are:

INPUT LEVEL - this selects that recording will begin once a threshold level has been exceeded. This is the default setting and one that is used by most people. The threshold is set in the REC2 page.

MIDI NOTE - this selects that recording will begin when any MIDI note is received. This is very useful when sampling a sound from a synth because the note on that makes the sound can also be used to start the recording.

FOOTSWITCH - this selects that the action of a footswitch closing will start recording. This may be useful for 'hands free' sampling. For example, let us

imagine you are sampling a heavy metal guitar though rather noisy amp and you are doing the playing yourself. Threshold based recording is no use because the background hum would set the sampler off. In this case, the footswitch may help.

monitor: Here you can select how you will monitor the signal you are sampling. There are two options: ON will select that as soon as you enter the REC2 page,the signal you are sampling will be heard 'through' the S3200. At the end of recording, it will automatically switch this 'through' signal out so you can hear your new sample. The other option, OFF, switches the through signal off completely (although this may be overridden in the REC2 page if you wish). This is used when monitoring the signal you are sampling through a mixer.

If you are sampling from a mixer it is possible to get 'feedback' if the S3200 is also connected to that mixer, set to ON and its channels are open..

(F)ree: This is not accessible but merely shows the amount of free memory. This can be displayed in sample points or milliseconds simply by pressing the REC1 key. The percentage of free memory is also displayed. The 'F' is in parentheses because this field is shown in other pages of the EDIT SAMPLE mode as an abbreviation.

bandwidth: This sets the bandwidth for the recording and you have two choices - 20kHZ and 10kHZ. Don't be put off by the 10kHZ option because you can make very respectable recordings at this sample rate. It is not possible to sample at any other rates but, if you wish, you may sample at 44.1kHZ (i.e. 20kHZ) and then re-sample it later to, say, 15kHZ to save on memory space.

orig. pitch: This sets the base pitch for the sound you are about to sample. At this point, you don't have to worry too much about this because you can set this parameter in the REC2 page and you can retune your sample in ED.2 later if necessary.

record tim: Here you may set the length of the sample you are about to make. The range is limited only by available memory space and whether the sample is stereo or mono. Again, you needn't worry about setting this field now as you can set it later in REC2 if you prefer. As you set this field, the fields below show you how much memory the new sample will take.

If you are not sure how long to set this, it is best to set it longer than you think you need. Samples can always be trimmed and edited later.

Once you have set the parameters as you like on this page, you don't have to worry about them for the rest of the sampling session as these are retained from sample to sample as you record them.

IMPORTANT NOTE - SAVING RECORD PARAMETERS TO DISK

It is possible to save your own set of record parameters to disk. By setting the parameters as you need them, save the operating system to floppy disk (go to disk, move the cursor to 'type of load:' and select OPERATING SYSTEM. Press SAVE and then WIPE and/or GO. This will save your personal record parameters to disk. From now on, when you boot up with this floppy in the drive, these record parameters will always be set for you. If you have a particular way of working, this may be invaluable and a great time saver. If you have several different ways of working, you might like to save different record set-up parameters to different disks and use the appropriate operating system as and when you need it. You will note that you cannot save several operating systems on one disk - you will need a separate disk for each system you save. Please refer to the DISK section for more information on saving files.

Having set your record parameters, the next screen to visit is REC2 so please press the key REC2. You will receive this screen display:

```
RECORD MONO V:LEFT  STRING C4       73%F
-20dB ptch:C_4 tim:  1.00s=  44100=   1%

SLCT REC1 REC2 ED.1 ED.2 ED.3 Mon  ARM
```

This shows you the main record page. Across the top of this screen are shown the type of recording about to be made (i.e. mono or stereo) and which channel of the stereo image you wish to see when making a stereo recording (i.e. LEFT or RIGHT). The name of the sample you are about to make is also shown. These can all be changed at this point if you wish. You may select STEREO for recording and you may select RIGHT for viewing (but only if stereo is selected). You may also scroll through the available samples should you wish to record over one. If no other samples are present, you may rename the one shown here by pressing NAME followed by ENTER.

***NOTE ABOUT NAMING SAMPLES IN REC2**: This is a temporary name with no specific copy function. If you go ahead and make the recording then a sample of that name is made but, should you select another sample, the new sample name will be lost. Please be careful so as to prevent accidental erasure of a sample.*

The remaining parameter on the top line is a percentage display showing how much memory is free.

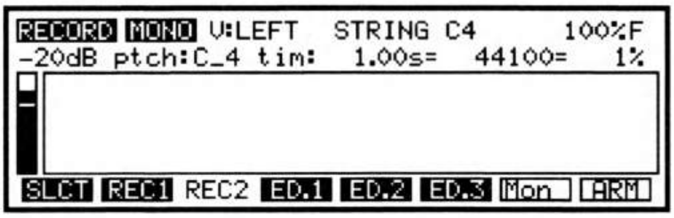
The next row of parameters allow you to set the threshold level, the base note of the new sample and the record time. Setting the threshold level is described below. You may set the base note of the sample directly from a MIDI keyboard (or other controller) if you wish. Providing you are monitoring through the S3200, pressing a note on the keyboard will set it here. You can, of course, set it manually using the DATA control. You may also adjust the length of time you wish to record for where it says `tim:`

You will note that many of the essential REC1 parameters are also on this page, albeit in an abbreviated way. This is done so that you can make recordings quickly with the minimum of fuss.

ADJUSTING THE RECORDING AND THRESHOLD LEVELS

Record levels are set by adjusting the front panel 'REC LEVEL' control. This works in conjunction with the three position GAIN switch on the rear panel. The level control should be adjusted so that the input meter to the left of the REC2 screen is as near to the top as possible.

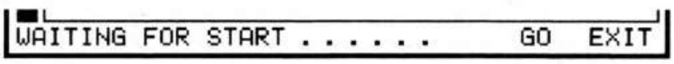
If you have selected INPUT LEVEL as the start mode in REC1, you will need to set the threshold level here. The default has been sensibly chosen for most purposes but you may find that some sounds with a slow attack are clipped slightly. To set the threshold, move the cursor to where it says -20dB. As you input your audio signal, set the threshold level accordingly so that it is low enough to catch the signal but not so low as to start recording on a false start. You will see a screen display such as:



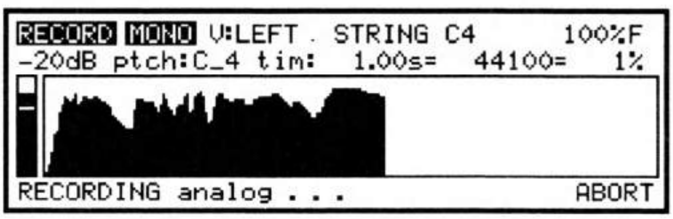
You will see the small level meter to the left of the screen bobbing up and down in accordance with the incoming audio signal and the threshold should be adjusted according so that the signal is just going slightly into the threshold box. It maybe that some adjustment of the front panel REC LEVEL control is also necessary. Once you have set the threshold level to your satisfaction, you are ready to make a recording.

RECORDING ON THE S3200

Assuming all your parameters are correctly set, all you need do is press ARM. You will receive the following prompt:



This indicates that either the threshold level has not been exceeded yet or, if MIDI NOTE is selected as the start type, that a MIDI note has not been received or, if FOOTSWITCH was selected as the start type, that the footswitch has not been pressed. You may also start manually by pressing GO. If you change your mind when this prompt is displayed (i.e. because you suddenly realise that continuing with the recording is going to overwrite a valuable existing sample), you may press EXIT. Assuming you press GO (or any of the start types get the thing going), you will receive this display:



The display will fill up with the sound as it is recorded. You may, at any time, abort the recording by pressing F8. This will finish the recording process early and abort that sample. This is useful if some mistake is made during the

record process or you were recording over another sample you wanted to keep.

After the recording has finished, the monitoring of the incoming signal through the S3200 is now automatically switched off allowing you to play the new sample directly from the keyboard or the front panel ENT/PLAY key. To switch the monitoring back on again, press Mon - METER ON. When you press Mon, the legend of that key changes to Moff (i.e. this key switches monitoring OFF). If you wish to take the sample again, there is no need to switch the meter back on again because pressing ARM will do this automatically. If, however, you wish to either listen to the source again (in the event of a mistake, perhaps) or set up for the next sample, specifically press Mon to switch the monitoring back on.

If you wish to make your next sample at this point, simply switch the monitoring back on again (press Mon), name the next sample and set a new base note if you need to (the easiest way is from the keyboard but the meter does have to be switched on for this - press Mon). Press ARM to start sampling.

RECORDING DIGITALLY

You may record digitally into the S3200 via the digital audio interface. This offers excellent sound quality with no deterioration of signal or increase in noise. If you own many sampling CD's (as distinct from CD ROM which is a different thing altogether), then you will find the digital interface very useful for creating top quality samples.

USING THE DIGITAL INTERFACE

The set-up parameters for the interface are found on F8 - DIGI. Pressing this key will display this screen:

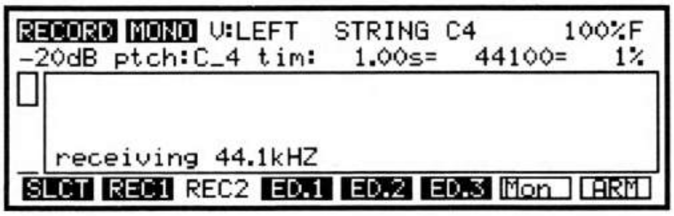
```
DIGITAL INTERFACE - Receive
  RECORD
        source:  ANALOG
         input:  ELECTRICAL
  receive rate:  AUTO

SLCT REC1 REC2 ED.1 ED.2 ED.3           DIGI
```

The fields are:

source:	This selects whether the digital audio will be received through the back panel jack or XLR analogue connectors (and hence via the S3200's analogue to digital converters) or via the digital inputs. The choices, you'll be surprised to know, are ANALOG or DIGITAL.
input:	This selects which of the digital inputs the audio will be received through if DIGITAL is selected in the field above. The choices are ELECTRICAL or OPTICAL which select whether the input will be received via the jack sockets or the optical fibre link.
receive rate:	This is not adjustable by the user. The S3200 works at a sampling frequency of 44.1 kHz only, not 48kHz.

***NOTE**: The parameters for the digital interface are also stored as part of the operating system (see above) so, if you always record digitally, then you might like to save the appropriate parameters to disk to save you having to set them up every time you wish to record.*

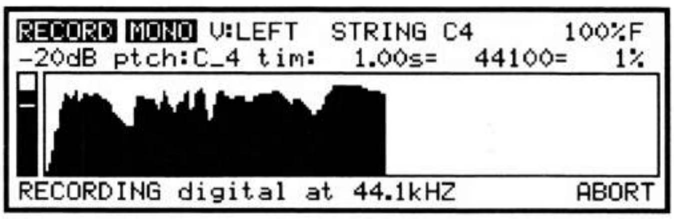
Assuming the interface has been set correctly, you will receive this screen when you enter REC2:



This is virtually identical to recording analogue except that you can see the message indicating that the S3200 is receiving digital audio correctly where it says `receiving 44.1kHZ`. If there is some problem with the digital audio signal, you will receive the following:

This indicates that the digital audio signal is not getting to the S3200. Please check your connections carefully - it could be that the cable has become disconnected. Alternatively, if your cables are properly connected, then please check the set-up parameters for the IB-302D as there could be something wrong there.

Assuming everything is connected properly and set up correctly, when you press ARM you will receive the usual `WAITING FOR START......` prompt and, if you press GO or when the threshold is exceeded, the MIDI note received or the footswitch pressed, you will get a display something like the following:



indicating the the S3200 is recording digitally. As with analogue recording, the display fills with the incoming sound's waveform.

Having successfully made your recording(s), analogue or digital, we can now move on to sample editing.

But first...

**** SAVE YOUR SAMPLES TO DISK NOW ****

It is good practice to repeatedly save your work as you go - all good programmers do this and it is a good habit to get into. You may make a mistake and accidentally record over a precious new sample, you may trip over the power cord and disconnect the mains, you may have a power cut - these things do happen!! Although saving is explained in detail in the section that describes the disk operations, to save you hunting for it, here is a quick explanation.

1. Insert a blank, formatted floppy disk or make sure your hard disk is connected and powered up.

2. Go to the DISK mode

3. If using a hard disk, select a suitable empty volume

4. Press SAVE

5. Press GO

The samples will be saved to disk and you can carry on working safe in the knowledge that, should something go wrong, you can retrieve it.

EDIT 1 - TRIM, LOOP AND JOIN

There are three main editing modes on the S3200. ED.1 gives you access to the basic editing functions such as TRIM, LOOP and JOIN. ED.2 gives access to the timestretch and re-sampling functions and ED.3 gives you access to the sectional editing, gain rescaling and sample fade function.

The first level of sample editing we have offers TRIM, LOOPING and JOIN. TRIM is where we can edit the start and end of samples, trimming out any unnecessary data. LOOP allows us to set up to four loops so that we can sustain short sounds. JOIN is where we can splice samples end to end, crossfade samples or merge samples. Our first port of call is TRIM.

TRIMMING SAMPLES

Pressing ED.1 in either the main SLCT or REC2 page will take you straight to the TRIM page and you will receive a display something like the following:

This shows the waveform of the selected sample (in the case of coming straight from REC2, this will be the sample you just recorded). You may edit the start: and end: fields accordingly.

The fields along the top allow you to select whether you want to edit in STEREO or MONO (it's pointless selecting STEREO if the sample is mono, by the way!) and you may, if you wish, select another sample for editing. The figure to the right of the top line shows how much memory is free.

Moving the cursor to the start: field allows you to move the start point. You will remember from the INTRODUCTION that you can change large

numbers such as this in several ways. You may type in a number directly from the numeric keyboard or you may use the DATA control. Each number field is separately accessible allowing you a great deal of flexibility in editing such large numbers. For example, to make a big change to the start point, move the cursor to the position just before the 1 - as you move the cursor you will increment in big jumps. This is good for getting someway into the sound with little effort. When you are close to where you want to be, move the cursor one position right to move in smaller increments. As you get closer and closer to the point you want to set, you can use finer incrementation until, with the cursor on the furthest right field, you are editing to a resolution of individual sample accuracy. All the time you are editing and changing values here (or in any sample editing fields), you can hear the results in real time by playing the keyboard.

Moving the start point will give a display such as:

You will see a vertical line indicating the position of the start point.

Moving the cursor to the `end:` field and adjusting that will give you a display such as:

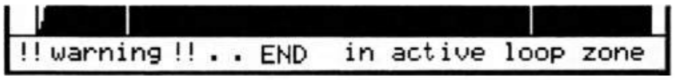
Playing the keyboard will, of course, play the edited sound. At this point, you may choose to destructively discard the portions either side of the start and end points by pressing CUT.

***NOTE**: This is a destructive process and, unless you have saved this sample to disk, pressing CUT here will irrevocably erase the data either side of the start and end points.*

Whilst it is possible to edit by ear with very good precision, it is sometimes desirable to see what you are doing. It is possible to zoom in on the waveform using the ZIN key. Repeatedly pressing this will enlarge the waveform display until you are looking at individual samples that are an inch wide! The display centres around the start point. Of course, now you can't see the end point so press the <--> key to switch between the start and end points for editing.

If the sound has had loops set previously, if you move the start or end point into one of those loops, the display will prompt you

or



If you attempt to use CUT at this point, it will be ignored so as not to upset good loops. When you get these messages, simply move the point in a different direction until they stop appearing - you are now out of the loop zone. You may now use CUT.

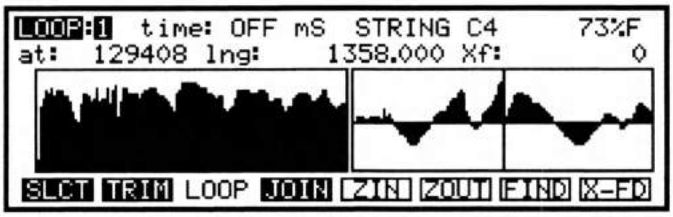
> ***NOTE****: To cut or not to cut, that is the question!!*
>
> *Whether you trim your samples and loop them afterwards or loop your samples and trim them afterwards is a matter for you to decide. Our programmers have found in their experience, that it is best to record the sample, loop it and then trim it afterwards. Of course, if there is 'dead space' either side of the sample, these should be removed before looping but it is felt that looping first and trimming later is safest.*

Let us now have a look at the LOOP page.

LOOPING

Looping is one of the trickiest areas of sampling but not impossible and the S3200 makes it as easy as it can with some functions to help you make good loops.

From the TRIM page, press the LOOP key to access the looping page. Again, you will see a display of the sample's waveform in the left part of the display together with a magnified display of the point where the loop rejoins the original sample sound. You can use the ZIN and ZOUT keys to zoom in or out of this window, but the display of the whole sample remains at a constant magnification.



The parameters are:

LOOP: Here you can select which loop you wish to set. The S3200 can have four loops for every sample. This may seem excessive but it can help to overcome the repetitiveness of some loops. You may select from 1 to 4 here.

> ***NOTES ON USING MULTIPLE LOOPS****: Whilst a sample may have multiple loops in it, there is one restriction you should be aware of and that is that they must be consecutive. That is, LOOP 1 must be followed by LOOP 2 , LOOP 3 and LOOP 4. You cannot have a situation where LOOP 2 is before LOOP 1 or LOOP 4 before LOOP 3 or LOOP 2. If you do set up such loops, certain ones will be ignored. For example, setting LOOP 3 before LOOPs 1 and 2 will cause LOOP 3 to be ignored. Please bear this in mind if you are creating multiple loops.*

`time:` This sets the length of the loop you wish to make. This is expressed in milliseconds. You can set a loop to last from 1mS to 9998mS. This is most useful when you have multiple loops. For example, LOOP 1 may be set to last for 5 seconds (5000mS), LOOP 2 for 1 second (1000mS) and LOOP 3 for 3 seconds (3000mS).

If you set 9999mS, this field changes to HOLD and when this is set, the loop will last as long as you hold a note.

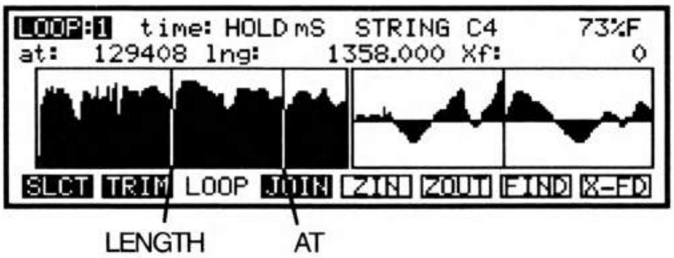
You may combine timed and held loops as you like. LOOP 1 may be set to last 1000mS, LOOP 2 set to last 200mS and LOOP 3 set to HOLD. When you play a note, regardless of where you are on the keyboard, the sound will play through LOOPS 1 and 2 over the time set and then move onto LOOP 3 which will sustain for as long as hold the note(s). This is a good way to breathe more life into a sample - single loops can sometimes sound repetitive if they are too obvious.

A loop will repeat for the greatest whole number of times possible within the loop time set here. For instance, if the total loop time is set to 250mS and the length of the loop is actually 175mS, the loop will only repeat once, not 1.428 times. This can save you a lot of calculation when you've set the loop length and you want a particular 'stutter' effect or repeated drum beat.

For the most part, you will probably find that one loop is sufficient for most sounds so don't worry about setting multiple loops for every sound - you don't *have* to use them.

***NOTE**: The easiest way to set HOLD is to type in 9999 - this will select HOLD.*

The name of the currently selected sample is shown next which can be changed if you wish. The memory percentage indicator is also shown. Below this line are the fields for setting the loop points.



`at:` Here you can select the point at which looping will begin. I.e. when playback reaches this point, it will go back to the point determined by the `lng:` field described below and will loop either for as long as the `time:` field is set or for as long as you hold the note(s) if `HOLD` is selected in the `time:` field.

lng: The actual length of the looped portion (as opposed to the length of time that the loop will repeat) is set in this field.

As you adjust these parameters, you will see two vertical lines move in the left part of the display, showing the position of the start and end points of the loop. You will note that this value is locked to the at: point - if you adjust the at: point, this parameter will also change. This is invaluable in cases where you have found a good loop length and want to re-position it elsewhere.

The length field is adjustable in very fine steps to allow you to manually set very accurate loops.

In the right half of the display you will see another waveform display. This is the point of the loop (i.e. the point set by the lng: parameter). As you change loop length or adjust the at: position, this area will display the waveform. The idea is to match these up as well as possible. You may use the ZIN and ZOUT keys to zoom in or out of this display for greater accuracy.

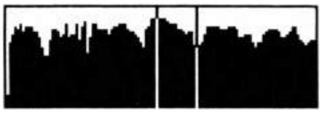
The FIND and X-FD keys are there to help you in your search for the perfect loop. The FIND key looks for points of equal amplitude. Repeated pressing of this will make the S3200 try and try again and you must judge if the loop is acceptable or not. The X-FD key is invaluable in creating loops. What this function does is to crossfade a portion of the sound before and after the loop according to the time set in the Xf: field next to the lng: field. This will smooth out any glitches you may have. This can be very effective in getting almost perfect loops. FIND and CROSSFADE functions are both non-realtime editing functions and may take a few seconds or more to calculate, depending on the length of the sample.

The FIND function does not work when looping stereo samples - this is because each side would have a different loop point and so go out of phase. It will only look for the best loop point on the currently shown sample.

NOTE: The crossfade function is destructive and will affect your sample permanently. Be sure to have saved your work before performing a crossfade in case you don't like the results or you make a mistake.

MAKING A GOOD LOOP

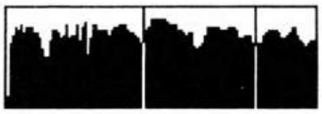
Making a loop is one of the trickiest things in sampling. The trick is to match two points in the sample that are similar in level and tone. For example, this would not make a good loop:



You have selected a high amplitude point in the sample and set a length that loops around a very quiet part. All manner of thumps and clicks would be heard and the loop would look something like this when played:

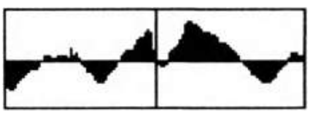
In this case, the window to the right of the loop page would look something like this:

You can see the abrupt level change. Setting these loop points may sound better however.



and the resulting sound when played would look like this

This is also a longer loop and so won't sound quite so obvious when held. The window to the right of the loop page would look something like this:



You can see a nice smooth transition.

Of course, there may well still be some glitches but this can be overcome using the crossfade key. Pressing that in this case may create something like the following:

Here, the crossfade function has smoothed the whole thing out to create a more uniform amplitude.

***NOTE**: Sometimes when crossfading, because part of the area chosen for crossfade may be slightly out of phase with the loop area, you may sometimes get a dip in level where they cancel each other out. This is not a fault of the S3200 but something that cannot be avoided.*

Using FIND and X-FD are probably your best allies in looping. FIND will look for good points of equal amplitude whilst X-FD will 'smudge' the whole thing to eliminate glitches, thumps and any other unpleasantness. Long samples work best with long crossfades whilst short loops are better suited to having short crossfades. Perfect results are not always possible but you'll be surprised at how easy looping can be on the S3200.

If your sample has just one single loop such as:

you can go to the trim page and remove the bit of audio after the loop as this will not be played. This can save a lot of memory space.

THE JOIN PAGE

In this page, you may join two samples end to end, crossfade them and merge them. This can be very interesting for creating long, evolving sounds and big, thick orchestral and synth textures. Pressing JOIN will display this screen:

```
JOIN  A then B-->J Free: 2326720=    73%
                      first    last    scale
A: STRING C4              0   25620   +00db
B: VOICES C4              0   34680   +00db
J: MIXTURE           X-fade over:      4350
- new name -              0 spl        0 mx
SLCT TRIM LOOP JOIN        A->J SPLI  MIX
```

The samples to be joined together are known here as A and B, and the resulting JOINed sample is called J. Though A and B can be the same sample, the resulting (J) sample cannot be the same as either A or B. Press the NAME key to enter a new sample name for the sample J. Existing samples can be overwritten and therefore can be used for J.

As mentioned earlier, this page has three basic functions: to copy a part of a sample to another sample - A->J - to splice a sample (or part of a sample) to another sample - SPLI - or to mix two samples (or parts of samples) together - MIX.

SPLICING SAMPLES

Select the A and B samples using the DATA control. You must then choose a name for the J sample. This can either be the name of an existing, unwanted sample, or you can enter a new name using the NAME key followed by pressing ENT/PLAY.

Now you should select the portions of the A and B samples which are to be combined using the `first:` and `last:` fields. For example, you may want to splice the just attack portion of sample A to the sustain portion of sample B, or mix just parts of two samples together. If you want to hear exactly what part of a sample you are going to use, you can set the first and last

points of A, and then press A->J - this will copy sample A into sample J without sample B so that you audition it in isolation using the ENT/PLAY key (you can overwrite J later, of course, so you've done no permanent damage to A if you get things wrong).

Notice how the figures at the bottom alter as the lengths of samples A and B are changed. The figure before the `spli:` is the total length of the selected portions of the two samples (minus the X-fade length - see below), and the figure before the `mx:` is the length of the longest sample portion to be included.

You may set the respective levels of each of the samples A and B using the `scale:` parameters but be careful you don't overload the system by setting too high a level.

Once you've set everything, press SPLI. If sample J already contains data, you will be asked if you want to overwrite it (you should press GO or ABORT as necessary). After a few seconds (depending on the length of the samples), you can hear the fruits of your labours by pressing the ENT/PLAY key.

CROSSFADING

To avoid a sharp break in sounds when you splice them together, one sound can be crossfaded into another. The crossfading will start before the value set in the `last:` point of A, the time at which crossfading starts depending on the number of samples set in the `Xfade over:` field. This field has no effect, of course, when layering samples.

Again, you can adjust the relative volumes of A and B by up to ±25dB, using the `scale:` parameters. However, if you set these too high, you may get a distorted sound, so use these with care.

When you've set up the start and end points for both samples, press SPLI. If sample J already contains data, you will be asked if you want to overwrite it (you should press GO or ABORT as necessary). Again, the process will take a few seconds (depending on the length of the samples and the amount of crossfading) and when the operation is complete, you can listen to the new sound by pressing the ENT/PLAY key.

MIXING AND LAYERING SAMPLES

It is also possible to 'stack' samples on top of each other. This can be an effective way of layering sounds without eating into polyphony. The same principles apply as above - select sample A and B and create a new sample J (or use an unwanted existing sample). You may set the portion you want to mix together (for example, you may want to layer the attack of one sound directly on top of another) and also set the levels and relative balance in the `scale:` field (please be especially careful here - because you are combining two samples together, the levels will naturally go up. If anything, you may need to use the scale parameters to turn them down). Once you're happy with everything press MIX. If the J sample already exists, you will be prompted and you should respond accordingly and, after a few seconds, you will be able to play the sound from the ENT/PLAY key.

You may repeat these techniques using the J sample as the basis for a new splice, crossfade or layer. For example, you could:

Crossfade a timpani strike with a deep string orchestra. Use that sample, for example, to crossfade with a big vocal chorus chord and use the result of that to crossfade with a large orchestral finale.

Crossfade a thick synth bass with a string pad. Use that to crossfade with a resonant filter sweep sound and crossfade the result of that with some strange percussion loop.

Layer some strings on top of a piano. Use the new sample as the basis onto which you layer a marimba.

Keep layering different synth string sounds on top of each other for a huge pad sound.

***NOTE 1**. When mixing, splicing or crossfading samples in this way, any loops which were present in the original samples (A and B) will not be played back when you play back J. You must reset loops in J if you want them.*

***NOTE 2**: The samples always use their base note (i.e. the note they were sampled at) when employing any of these techniques. In other words, a string sound sampled at C3 and layered or crossfaded with a choral sound sampled at G3 would be a fifth apart. This cannot be corrected, even through tuning it in ED.2 (see later).*

***NOTE 3**: It goes without saying that you need to have sufficient memory available to create these new spliced or mixed samples. You will be reminded by the prompts if you haven't!*

Though the process of creating the sound you want may take some time, it's possible you may discover some new sounds along the way which weren't quite what you were expecting, but could find a place in your work. The type of sounds created here can be long evolving sounds which may find a place in soundtrack work as they can be very dramatic.

Let us now have a look at some of the possibilities offered in ED.2.

EDIT 2 - TIMESTRETCH AND RE-SAMPLE

The ED.2 page (accessed from the SLCT or JOIN page) allows you to perform some further sophisticated editing functions such as Timestretch and re-sampling.

THE PARAMETER PAGE - TUNING AND REVERSING SAMPLES

The first page we see is the PARAMETERS page (the first one you access when you press the ED.2 key) and looks like this:

```
PARAMETERS of sample: STRING C4          73%F

  original pitch: C_4
     pitch offset:+00.00    (semi.cent)
type of playback: LOOP IN RELEASE
loop tune offset: +00 cents (HOLD only)
SLCT PARA TIME RATE                       REV
```

On the top line is the name of the sample you want to edit. As usual, this may be changed with the DATA control. The parameters on this page are as follows:

`original pitch:` This allows you to alter the original pitch at which the sample was recorded, so that when you replay it on the keyboard, it will play at the correct pitch. You will note that the ENT/PLAY key which normally plays a sound at a pitch set in the MIDI TRAN(smit) page, plays the sample at its correct pitch (i.e. the pitch it was sampled at) when you are in EDIT SAMPLE.

`pitch offset:` Further fine tuning (in semitones and cents over a range of ±50 semitones) is possible here.

`type of playback:` This parameter determines the way in which the sample will be played back. There are four options available here.

The first is `LOOP IN RELEASE.` This means that when a key is pressed, the sample will play through all the loops until the first HOLD loop is reached. When the key is released, the HOLD loop will continue to play as the release falls away. This is always selected when you make a sample.

`LOOP UNTIL RELEASE` is slightly different. Again, the sample will play, with all loops, until the first HOLD loop is reached. However, when the key is released, the loop will end, and the remaining portion of the sample (if any) will be played. This is a useful setting for sounds which have a definite attack, an indefinite sustain period (set with the loop), and a definite release characteristic. For example, if you have a double bass sample that has an interesting and realistic finger squeak at the end, this type of loop may be appropriate.

`NO LOOPING` does what its name suggests - it plays the sample through without loops for as long as the key is held down. If the sound is not long enough, it will finish even though you are holding down a note. If the sample is still sounding, as soon as the key is released, the sound will start to decay.

`PLAY TO SAMPLE END` is useful for triggering sampled drums either from a keyboard or sequencer or by drum trigger units such as the ME-35T and the like. Like the previous setting, no loops are played, but an instantaneous trigger signal or key press will play the whole of the sample (the key does not have to be held down for the whole length of the sample).

NOTE*: Alternating backwards/forwards loop is not available. This is due to hardware and not software. To have included it would have meant the loss of other features such as polyphony or resonance on the filters. It cannot be included in future software upgrades*

`loop tune offset:` This allows you set a pitch shift of up to ±50 cents (one semitone) for a HOLD loop. This function is useful when you have a small, short, single cycle loop that has

latched onto some strange, discordant aspect of the sound and is slightly out of tune with the rest of it.

REVERSING SAMPLES

In the ED.2 PARAMETERS page, when the REV soft key is pressed, the sample will be reversed. Pressing it again will reverse the sample back to its original form. Note that any loop points you have set will stay in the same absolute positions, and will not be reversed with the sample.

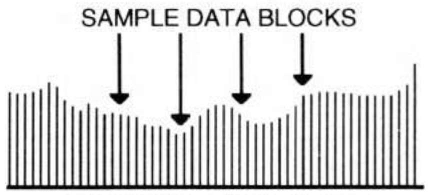
TIMESTRETCH

One of the most useful digital signal processing techniques available today is that of time compression and expansion or 'Timestretch' as we call it at Akai. This technique allows you to alter the length of a sample, shortening it or lengthening it, without changing its pitch, the common problem of trying to change the length of the sample by playing it back slower or faster.

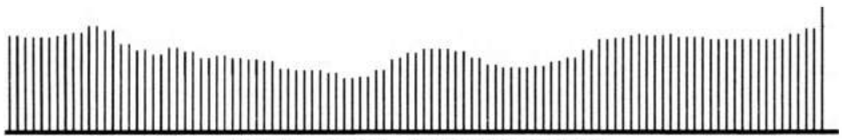
Before we look at how to operate the timestretch functions on the S3200, let us first look briefly at how timestretch works.

Timestretch works by instructing the digital signal processor to analyse the signal and insert or delete blocks of sample data at appropriate places and crossfades are used to make the insertions and deletions as seamless as possible. This has the effect of lengthening or shortening a recording.

As you can see from the following diagram, blocks of sample data have been inserted to create a 200% timestretch. The overall envelope of the sound data is preserved but there is twice as much data in it causing it to play back twice as slow.



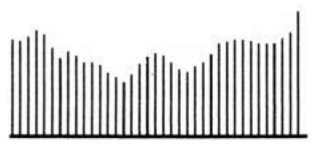
ORIGINAL RECORDING



WITH 200% TIMESTRETCH

In the following diagram, data has been carefully removed to make the recording play back faster.

ORIGINAL RECORDING



WITH 70% TIMESTRETCH

You can see that the length in both of these examples changes quite drastically but the envelope remains pretty much the same, thereby ensuring the

integrity of the sound. The timestretch on the S3200 works by making a copy of the original. It is necessary to make a copy for two reasons - firstly, the S3200 needs the original data to get the data to insert. Secondly, in the event of a stretch going wrong, you still have the original to revert to.

Unfortunately, however, perfect results using timestretch are sometimes difficult to achieve. It is not a limitation of software or hardware but due to the fact that, although the processor is clever, it is not *that* clever and can sometimes make mistakes in deciding which sample to insert or remove. The end result of this is that, on occasions, especially with stretch factors exceeding 10% or so, you may get an echo or 'flam' effect on some transients because the processor has inserted a transient. When shrinking a recording, you may find a transient softened because the processor has decided to remove it. You will find this to be the case more or less on all devices that feature some form of time compression or expansion.

A lot of these problems depend on the nature of the audio material being processed and settings that process the spoken word perfectly could make a right mess of a percussive dance track. The converse is also true. The biggest problem is in material that has a healthy balance of low and high frequencies because different timestretch parameters are required to process different frequency ranges - in audio material that has a wide frequency composition there is much adjustment to be done to obtain the correct compromise so that both frequency ranges are adversely affected as little as possible. Please be aware that, on occasions, you may never get absolutely perfect results and there may be occasional side effects, especially with extreme settings of stretch - of course, these side effects can be put to good use for the creation of special effects!! Over smaller ranges, however, you will find the timestretch on the S3200 yields excellent results and will become an invaluable tool in your work, whatever application you are working in.

You could use timestretch to alter the length of a drumloop or breakbeat without altering the pitch to fit in with the rest of the track or you could use it to speed up or slow down a track to change the 'groove' or feel. You could even use the timestretch to overcome timing discrepancies of a 'live' band or use it to create gradual tempo changes, etc.. It can be used to change the length of, say, a backing vocal part or guitar solo so that it can be played back at a different pitch and so accommodate key changes. You could use it to maintain the same vibrato speed for a number of 'multi-samples' created from one original sample. In A/V work, it could be used for fitting sound effects, voice overs, etc., to video soundtracks for precise timing. In radio broadcast you will be able to process voice-overs to precisely fit jingles (or jingles to fit voice-overs!) or to process jingles to accurately fill timed advert slots. There are many possibilities which you will no doubt discover for yourself.

TIMESTRETCH PARAMETERS

Let us now examine how to operate the Timestretch function on the S3200. Pressing the TIME key from the ED.2 page enters the TIME-STRETCH page.

```
TIME-STRETCH  sample: STRING C4     73%F
stretch zone:        0    to:      128
Cycle length: 1000   total:  220512  7%
 time factor:  100%  norm. time= 5.00sec
stretch mode: CYCLIC  qual: 10 width: 10
  new sample: STRING C4  *existing Samp*
SLCT PARA TIME RATE autC ZONE  GO  PLAY
```

This enables you to lengthen or shorten a sample or a selected part of a sample from 25% of its original length to 2000% (twenty times) without changing its pitch.

Two modes are available for stretching: CYCLIC, in which a fixed interpolation rate is maintained throughout the whole of the sample (suitable for individual instrument samples), and INTELL, in which the S3200 'intelligently' varies the interpolation rate according to the sample content (suitable for speech and music).

As usual, you may select the sample to be edited at the top of the page. The parameters on this page are:

stretch zone: Here you may set two values to set the start and end of the area of the chosen sample you wish to stretch. You may only want to stretch one part of the sample so it should be set here. The first field sets the start point of the stretched area and the to: field sets the end. You may audition the area you have set using the ZONE key.

Cycle length: Here you can set the cycle length (in samples). The soft key autC can be used to help you find the right sample length. As with autolooping, the S3200 applies software logic to the sample to calculate what it believes is the right answer but, like autolooping, whilst the S3200 will often help you, it is not always infallible. The Cycle length: function only applies to the CYCLIC mode if timestretch.

time factor: This sets the percentage by which the sample will be stretched or shrunk. The range is 25% to 2000% (although we are the first to admit that such extremes are only going find favour with the truly mad!). As this parameter is adjusted, you will see the length of the sample changing in the adjacent norm.time: field and in the total: field above that.

stretch mode: There are two ways in which you can stretch a sample and this is selected here. CYCLIC uses a fixed cycle time at which the S3200 will stretch. When INTELL is selected here, the S3200 makes its own decisions as it proceeds with the stretching process. Be warned, though! Although the intelligent mode will produce better results, the time taken for this operation is much longer than when the CYCLIC mode (up to several minutes depending on the length of the sample and the amount of stretch).

Remember that to perform any of these operations, you will need to have enough free memory.

qual: This sets a level of intelligence for the S3200 to work with when performing an intelligent timestretch. It sets the number of decisions it will make as it works its way through the sample. With lower values set here, it will not make so many examinations of the sound. With higher settings, it will examine the sound in great detail and so produce better results although

this will take longer. This control only has a function when INTELL is selected.

width: This sets a crossfade between the original and the inserted data. It is recommended that when low qual: values are set, this should be set high and vice versa. This control only has a function when INTELL is selected.

PERFORMING A TIMESTRETCH

Set the parameters as described above - if you are stretching complex samples such as breakbeats, backing tracks, drumloops, voice over, backing vocals, etc., use the INTELL mode of operation. For stretching individual instrumental samples, maybe CYCLIC will be o.k..

First set the zone you want to stretch in the stretch zone: and to: fields. If you want to stretch the whole sample (which is the usual application), this field will be set as soon as you select the sample.

If you are using CYCLIC mode, then set a cycle length (or use the autC key). If you are using the INTELL mode, set the qual: and width: as you think necessary (remembering that high qual: values will take more time). Now set the stretch factor: parameter, name the new sample and press GO.

If you haven't created a new sample, you will receive this prompt:

```
can't replace source sample
```

In this case, please name a new sample or select a sample you know you have no further use for as the destination sample.

Whilst the timestretch is processing, you will receive the following display:

```
**** BUSY - PLEASE WAIT ****
```

Depending on the mode you selected and the settings of the parameters, you may have to wait a while (several minutes in the case of long INTELL stretches). You can abort the procedure by pressing F8 a few times. You will see the display counts down in percentage the time remaining for the process.

When the processing is finished, you can play back the original from the ENT/PLAY key or the stretched version from the PLAY key. If you are happy with the sound of the stretched sample, you can proceed to edit, trim and loop it, just as if it was a freshly-recorded sample.

RE-SAMPLING

The re-sampling page (called RATE on the soft key because we didn't have enough room!) allows you to re-sample your sounds at different sampling rates. This is to allow you to sample at anything other than 44.1kHz and 22.050kHz and to save memory. High sampling rates and bandwidths are all well and good but if the sound doesn't have a high harmonic content, what's the point of wasting valuable memory? For example, you would be justified in using a high sampling rate for cymbals and hi-hats with their high harmonic content but for bass drums, toms, amplified electric guitars and the like

whose frequencies may not extend beyond 10kHz, it seems pointless. Of course, this kind of sound can be set to be recorded at 10kHz in the REC1 pages but what if the sound needs a bandwidth of 12 or 14kHz? This is where the re-sampling or RATE page comes in.

Pressing RATE in ED.2 calls up this screen:

```
RE-SAMPLE      sample: STRING C4      73%F
 present sample rate: 44100 Hz
     new sample rate: 22050 Hz
          new length:   110256     = 4%
         tune offset:-12.00 semi.cent
  new sample: STRING C4   *existing Samp*
SLCT PARA TIME RATE 3/4 2/3  GO  PLAY
```

As usual, the sample name and free memory are displayed on the top line.

`present sample rate:` This shows the selected sample's sample rate. This field is not accessible.

`new sample rate:` This allows you to set the sample rate of the new sample you wish to create. This is variable between 22050Hz (22.050kHz) and 65000Hz (65kHz). There may seem little point in re-sampling upwards but it might come in useful if you need to transfer a sample via a sample editor to a sampler that uses a higher sample rate. The default for this field is 22050 (half bandwidth) but can be set as you like. This field also works in conjunction with the 3/4 and 2/3 which enter three quarters and two thirds the sample's original sample rate.

`new length:` This shows the length of the new sample. As usual, as in all pages of EDIT SAMPLE, you may view this in milliseconds by pressing the RATE key again.

`tune offset:` This shows the new tuning. When sounds are re-sampled, they must adopt a new tuning. This is because, when you re-sample at, say, half bandwidth, you take out half the data making it play at twice the pitch so it is necessary to offset the tuning. This is done automatically for you and this field is not accessible.

PERFORMING A RE-SAMPLE

First, create a new sample - re-sampling is a copy process and so you have to create a new sample by pressing NAME, entering a new name and pressing ENT. If you wish to use either of the two preset sampling rates, simply press either 3/4 or 2/3 - this will re-sample the original to three quarters or two thirds the original frequency. Alternatively, set a value of your choosing in the `new sample rate:` field.

If you have not named the new sample you will receive the prompt:

```
can't replace source sample
```

and you should input a unique name. During the re-sampling process (which is very quick but does depend on the length of the sample) you will receive the message:

```
**** BUSY - PLEASE WAIT ****
```

When the process is complete, you can play the new sample by pressing the PLAY key.

You can use the re-sampling facility to save on memory. In the studio where you have time to load in new sounds this may not be so crucial but on-stage, you ideally need to cram as much into the S3200 as possible. Even with 32 Megabytes of RAM installed, there will be occasions where you need to squeeze that little bit more out of the samplers internal memory. In this respect, the re-sampling functions are ideal - in fact, given that listening conditions at a gig are nowhere near as critical as in the studio, you could afford to make special 'gig disks' where the bandwidth is more limited than it could be to save even more space in your memory.

EDIT 3 - SECTIONAL EDITING, NORMALISATION, DIGITAL FADES

ED.3 introduces new functions not previously found on the S1000/S1100. Existing owners of these samplers are forgiven if you have not read the previous section because the functions are virtually identical to the S1000 and S1100. This next section will be of benefit to new and old users alike. These new editing features include sectional editing, level rescaling and normalisation and digital fades.

Pressing ED.3 in the main SLCT page will take you to this screen:

This is the SECTIONAL editing page and shows a typical drum riff, something for which the sectional editing functions are ideally suited for editing. Whereas TRIM allows you to remove audio data either side of the start and end points, these functions allow you to remove data *within* the start and end points. This can be used, for example, to remove one single snare beat from a drumloop sample or, in audio/visual applications, could be used to remove a mistake or a cough in a piece of dialogue. You may extract such pieces of audio in several ways. You may extract it and leave the gap it creates or my extract it and close the gap it creates. You may overwrite the existing sample or you may extract the section to another new sample, keeping the original sample intact. There are many uses for these new functions and you will no doubt find your own.

To create a sectional edit, set the start and end points as appropriate. You may use the ZIN and ZOUT and <--> keys as in TRIM to get more accurate edits. As you set the start and end points, you will receive a screen display such as:

At this point, you may like to create a name for the new sample to be copied to. You don't have to as it is possible to overwrite a sample with the edited version but, if you want to be safe, it's probably best to make a copy, assuming you have enough available memory.

Now press EXEC. You will receive this screen:

Here you are presented with four choices. These offer three distinctly different edits depending on what it is you want to achieve. They are:

exit — This will exit this screen and return you to the main SECTIONAL edit screen without having any effect on the sample.

EXTR — This is an 'extract' function that will remove the isolated area and copy it across to the new sample. I.e:

This is particularly useful for isolating such things as single snare or bass drums from a pattern although any sound could be 'lifted' in this way.

CUT — This allows you cut the section you have marked and keep the gap thus created. I.e:

This is good for removing offending noises from a track where the rhythm or pace should be retained. For example, in a vocal line where the singer accidentally knocked the mic stand or sneezed! It is also good in

dialogue where you want to remove a cough or script pages turning.

CHOP — This will remove the selected area and close the gap thus created. I.e.:

This is good where you want to remove something but the rhythm or the pace is not so important.

If you do not name a new sample, you will receive the following prompt when you press EXEC:

```
overwrite existing sample?        GO  ABORT
```

Pressing GO will take you directly to the `select:` prompt and you may EXTR, CUT or CHOP as you wish. If you change your mind, press exit.

Whilst the S3200 is processing the data, you will receive a 'busy' prompt. After a few seconds, the new sample will be shown which may be played in the usual way.

As with TRIM, the S3200 is very considerate and won't let you ruin good loops and you will receive the warnings:

```
!! warning !! . .START in active loop zone
```

or

```
!! warning !! . . END  in active loop zone
```

In this case, whatever editing action you perform will be ignored.

As with Timestretch and Re-sampling, if you create a new sample from any of the above editing procedures, the original loop points will be lost and you will need to reset them.

You may select to edit in mono or stereo accordingly. Of course, this only has relevance if the sample is stereo in which case you may prefer to edit the left and right channels separately.

***IMPORTANT NOTE ABOUT SETTING START AND END POINTS**: The start and end points are temporary whilst in ED.3. They are not stored anywhere. If you leave ED.3 or select another sample, these will be lost and you will have to set them again.*

LEVEL SCALING/NORMALISATION

Pressing SCAL in any of the ED.3 pages will display this screen:

Here you may set new levels for the selected sample and also normalise it.

Normalising is a technique where you can analyse the loudest part of a signal and then adjust the whole sound to be at its optimum level - i.e. the loudest part of the signal is at the unit's maximum level before distortion and the rest of the sound is brought up in level proportionately. This increases signal to noise ratio and dynamic range. It also allows you to compensate for recordings made at too low a level. This page also allows you to set your own level for gain rescaling if you wish. You may process in mono or stereo (although selecting stereo is pretty pointless if it's a mono sample!)

To normalise a sound, simply press NORM. If you have not named a new sample for the normalised version to be copied to, you will receive the prompt:

```
overwrite existing sample?      GO  ABORT
```

to which you must respond accordingly by pressing GO or ABORT. After a 'BUSY' message, the new sample will be displayed showing the new level.

To rescale a sample to a level of your own choosing, simply move the cursor to the rescale value: field, enter a value of your choosing and press RSCL. Again, you will receive the usual prompts if you haven't created a new sample to copy to and after a few seconds, you will see the sample scaled to the value you set.

***NOTE 1**: Please be careful when using RESCALE because it is possible to overload the unit if you set this value too high.*

***NOTE 2**: Please also note that whilst normalise and rescale can save the day on many occasions when you have recorded a sample at too low a level, it can increase noise levels. For example, if the sample is very quiet, as you boost the level, you will also boost the noise floor. Setting, for example, a rescale value of +15dB to bring the level up to maximum will also boost the noise level by 15dB. Please bear this in mind when using these functions.*

SETTING DIGITAL FADES

It is also possible to set fades on a sample. This may seem unnecessary seeing that you may effectively set fades using the envelope generators in EDIT PROGRAM but it does have uses. Probably the one that springs to mind first is where you have a noisy drum sample. Whilst you could 'shape' the noise out in EDIT PROGRAM using envelope generators, it means that you have to do this every time you want to use this particular sample. By setting a digital fade down, you can affect the sound at source.

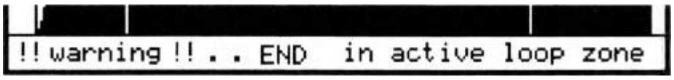
Pressing FADE in any of the ED.3 pages will display this screen.

Here we see a string sample and a start and end time has been set. Pressing GO will give you the usual prompts if you have not created a new sample to copy to. If you agree to overwriting the original sample or have created a new one, after a few seconds you will receive a display not unlike this one:

Here, you can see the sample has fades which lead up to the points set by the start and end position set above. You may play the sample from your keyboard or the ENT/PLAY key.

If you try to set fade times that fall within any loop zone(s), you will receive the following warning:

or



In this case, whatever editing action you perform will be ignored.

One thing to remember when setting fades is that these will speed up and slow down as you play them across the keyboard range so, while it may seem a good idea at the time to set a slow legato fade up and down on a string sample, you may find it would have been better to have used an envelope generator so that attack and release is consistent across the keyboard range.

***IMPORTANT NOTE ABOUT SETTING START AND END POINTS**: The start and end points are temporary whilst in ED.3. They are not stored anywhere. If you leave ED.3 or select another sample, these will be lost and you will have to set them again.*

Don't forget that, if you wish, you may reference your display in milliseconds by pressing FADE again. This will show the fade time as it affects the sample at its base pitch. Please remember that this is for display only - you cannot edit in milliseconds.

CONCLUSION

As you can see, there is a lot one can do to modify a sample once you have it in memory. But the fun is only really beginning. In the next section we will see how we can map these edited sample out across the keyboard as well as discover many other interesting possibilities.

EDIT PROGRAM

The EDIT PROGRAM mode is where you assemble your raw, edited samples for playback. In the EDIT SAMPLE mode, the samples are unprocessed by envelopes, vibrato, etc.. If they have been looped, it is quite possible they have lost all their dynamics - this can be overcome in EDIT PROGRAM. Furthermore, because of powerful synthesizer functions, the S3200 can be used to play and process samples much like an analogue synth. With 2 low frequency oscillators (LFOs), ADSR and multi-stage envelope generators, two banks of resonant filters, panning and more, the S3200 can radically transform any sound offering the creative musician and programmer endless possibilities.

At the heart of EDIT PROGRAM is the ASSIGNABLE PROGRAM MODULATION or APM for short. This allows sophisticated modulation of modules in a freely assignable manner and virtually any modulation source may be assigned to almost any sound processor in mixable and invertable amounts. If all that means nothing to you, don't worry for the moment as we will cover it in depth in this section. In short, what it means is that as well as owning a superb sampler, you also have a very excellent and versatile analogue style synthesizer.

On top of this, you may set sophisticated keyboard splits and layers, set velocity switching and crossfading, assign samples to individual outputs and/or pan them in the stereo outputs as well as tune and transpose your samples. You may also set MIDI parameters for your sample(s).

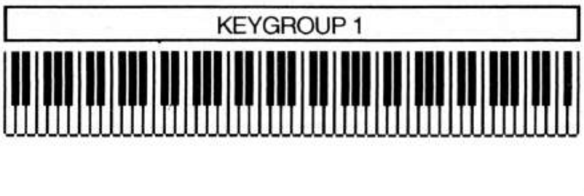
"But", you may be asking "why have programs? Why not just play samples from EDIT SAMPLE?" A good question. The reason we have EDIT PROGRAM is because raw samples, however much you may have trimmed, looped, crossfaded, stretched, etc., are only half the story. With samples, it is necessary to assign them to different areas of the keyboard for playback - in EDIT SAMPLE, you can only play one sample at a time spanning the entire keyboard range.

WHAT IS A PROGRAM?

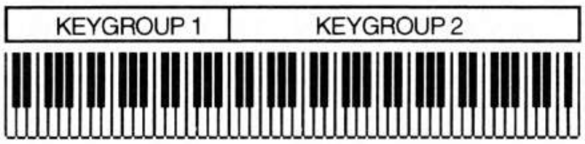
In the last section, we dealt with multi-sampling. This is the technique where you take several samples of one instrument across its range or where you take several samples of different instruments (such as drums, for example). Aside from all the fun things you can do with envelope shaping, filtering and so forth, it is in PROGRAM EDIT that you map all these multi-samples across the keyboard. To do this, we place the samples into what we call KEYGROUPS.

WHAT IS A KEYGROUP?

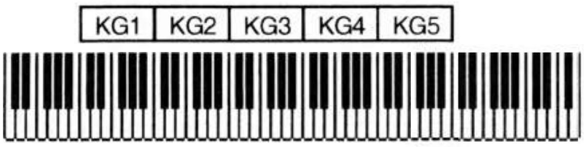
A KEYGROUP is precisely that - a group of keys which have a particular note range on the keyboard. The simplest program you can have is with one keygroup in it that spans the entire MIDI range on C0-G8. The TEST PROGRAM that always boots up into the S3200 is just such a program. I.e:



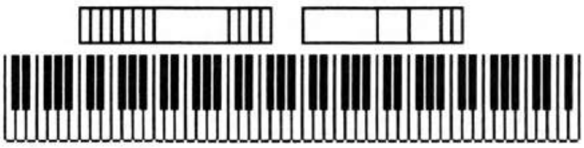
Perhaps the next level up is to have a program with two keygroups. One covers the range C0-B2, the other C3-G8 - this would be a simple keyboard split. I.e:



The next level may be a program which has five keygroups - one for each octave on a normal synth keyboard. Such a program may be useful for something like piano or strings which have been sampled on the G of every octave. I.e:



After that, of course, it's anyone's guess what the next level may be but it could be something like this:

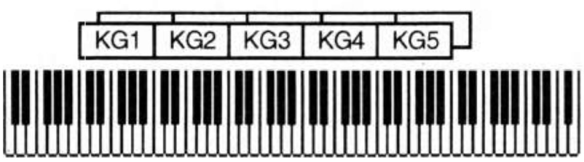


Here we have 17 keygroups in one program - some are assigned to individual keys whilst some span an octave or so. Hopefully, you can now get a feel for how flexible the keygroup assignment can be. But there can be more to it than that.

KEYGROUP ZONES

Within each keygroup, you may assign up to four samples in what are referred to as ZONES. These can be used for a number of things that include velocity switching and crossfading, playback of stereo samples and layering.

To playback stereo samples or to layer sounds or just to do a simple velocity switch/xfade, you could have something like this:



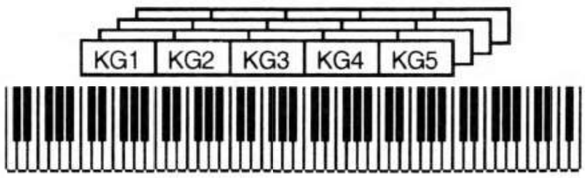
Here we have five keygroups, each with two zones being used. You could use this type of assignment for playing back stereo samples where the left and right samples are assigned to their own zones (1 and 2 respectively) in one keygroup and each zone is panned hard left and hard right. As an example, you could have five stereo string samples each sampled at G on every octave which could be assigned into such a program.

This kind of program is also suitable for layering sounds on top of each other. In the above example, four synth samples could be mapped out on the keyboard (one for each octave, perhaps) and zone 2 of each keygroup could

also contain the same sample as zone 1 and these could be panned and detuned for a fat, pseudo stereo layered synth sound. Of course, each zone could have different synth sounds in them.

This type of program is also suitable for a two way velocity switch - by setting zone 1 of each keygroup to a velocity range of 0-90 and setting zone 2 to 91-127, you could, for example switch between a thumbed bass and a slapped bass or a normal snare hit and a rim shot using velocity.

To round things off, each keygroup has no less than four zones and a program may 'look' something like this:



Here, we have five keygroups, each using the four zones. This may be for a four way velocity switch or for velocity switching between two stereo samples or for layering four samples together.

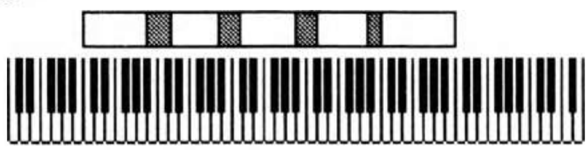
The ultimate program, would be for each key to have its own keygroup with each keygroup containing four samples using a four way velocity switch!

OVERLAPPING AND CROSSFADING KEYGROUPS

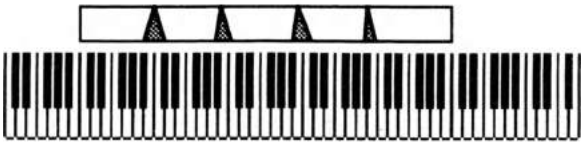
So far we have seen keygroups side by side. This is usually fine for most applications but there are sometimes occasions where the abrupt transition between one keygroup and another can be a bit obvious. For example, in a strings program where you have five string samples each at the G of every octave, the transition between B2 and C3 may sound a little strange.

The reason for this will normally be that the G2 sample is transposed up by four semitones at B2 and so sounds a little brighter whilst the G3 sample is being played 7 semitones down so it may sound a little duller and so, next to each other, especially when playing a scale, the crossover point is not even.

To overcome this, we can overlap keygroups simply by setting their key ranges accordingly:



It may be, however, that this does still not overcome the problem and so there is a facility to crossfade keygroups for an even smoother transition where one keygroup gradually fades down through the overlap whilst the other fades up thereby giving a smooth transition. I.e:



Of course, you can use a combination of any of the above techniques and have crossfading, velocity switched keygroups in programs alongside layered and

split keyboard assignments. The above diagram examples represent only part of the flexible program editing and multi-sampling potential of the S3200.

If all this seems very confusing, don't worry for the moment. There are many easy routines in EDIT PROG that allow you to edit all keygroups simultaneously or to copy keygroups. Key ranges can be conveniently set by playing your MIDI keyboard if you wish and, within a short time, you will be making programs very quickly and easily.

If you have already owned an S1000 or S1100, then you will probably be familiar with a lot of what we have just seen but, if this is your first time with an Akai sampler, it is worth taking the time to get a fairly good understanding of these principles and the concept of keygroups if you are going to get the best out your sampler.

But before we can move onto to see what making up a program is all about, we need to have a look at another important aspect of a program - ASSIGNABLE PROGRAM MODULATION or APM.

ASSIGNABLE PROGRAM MODULATION

APM is a new concept in sample editing. Many synthesizers have had such facilities before but this is the first time such a concept has been introduced on a sampler. APM turns the S3200 into a powerful synthesizer as well as offering a great deal of flexibility in the manipulation of acoustic samples.

In the early days of synthesizers, each building block of sound was referred to as a 'module' and it was possible to route any module to any other. This was called 'modular synthesis'. Of course, in the wrong hands, all manner of tastelessness ensued but this style of synthesis was both expressive, intuitive and versatile. Since those days, with the advent of digital synthesis, such flexibility has all but gone (although, in fairness, we have had enormous benefits in other ways). Now, on the S3200, many of the functions found on those classic synths are available again.

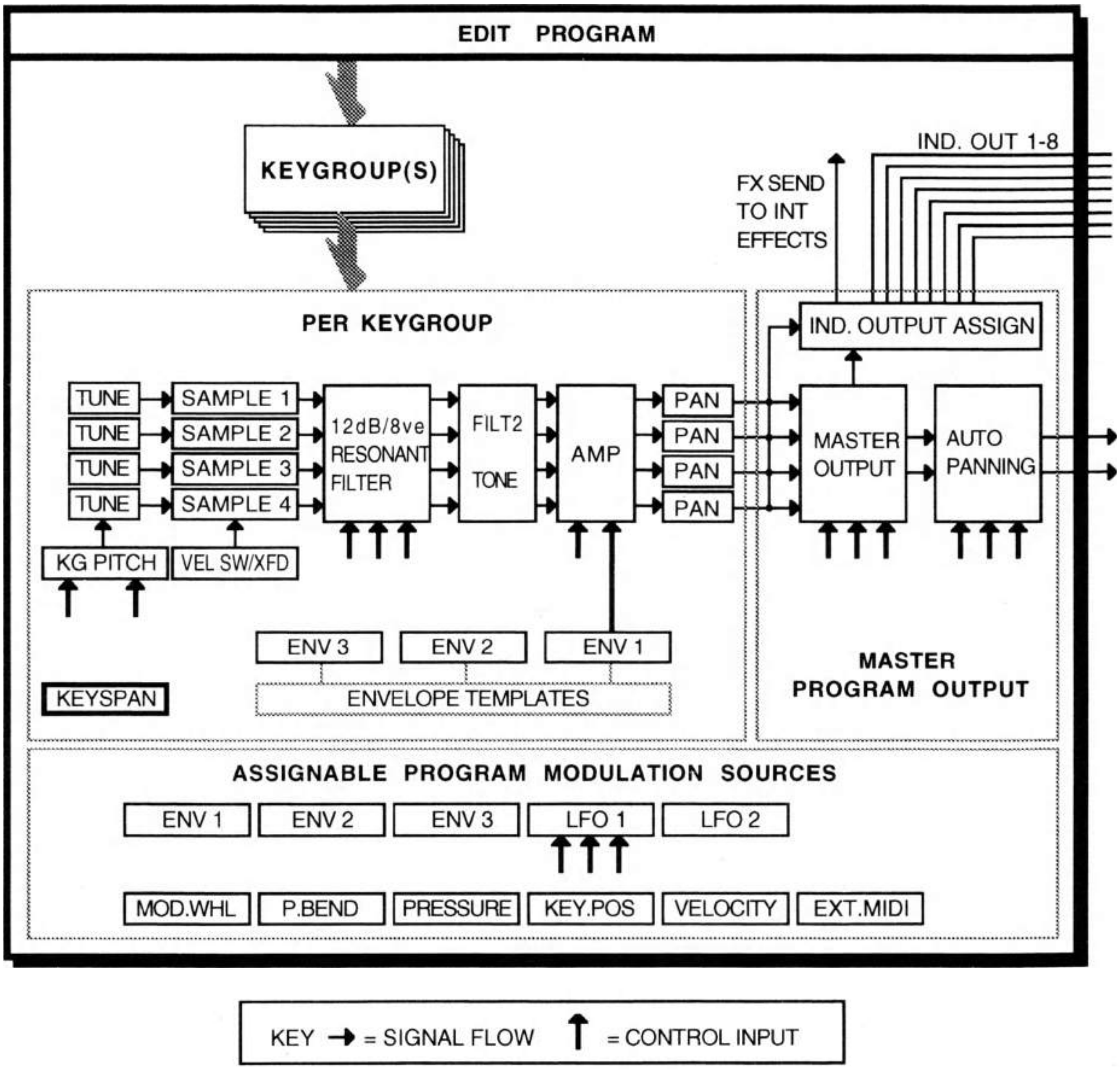
On the S3200, each module (i.e. the filters, amplifiers, pitch inputs, LFO's, envelope generators, etc.), have several control inputs. On other samplers (and indeed even some synthesizers), these control inputs are fixed - that is, you have no choice as what you can send to them. Whilst acceptable a lot of the time, this can be a bit frustrating when there is some specific sound you want to make or musical effect you want to achieve.

On the S3200, such control assignments are freely variable to be set by you and you may route virtually any controller (including MIDI controllers) to virtually any module. This means that, for example, you may route the mod wheel to open the filter in a brass program for swells and growls or you may route aftertouch to control the panning LFO speed to emulate the sound of a rotary speaker speeding up and slowing down in a classic rock organ sound. The multi-stage ENV(elope) 2 could be routed to pitch and inverted for special effects whilst, at the same time LFO1 (whose rate may be separately controlled by LFO 2, for example) is routed to a resonant filter cutoff. You could use ENV(elope) 1 to control LFO1 depth for 'shaped' vibrato - all sorts of things are possible from the subtle to the ridiculous! We are the first to admit that this kind of modulation is not that useful when trying to accurately recreate the sound of a Gamelan orchestra but, when the need arises to create that special sound or when your synthesizer just isn't up to it, the S3200 will oblige you willingly - in fact, you may get so used to the S3200's

versatility as a synthesizer that some of your current synth(s) may be in the classifieds sooner than you think!

This block diagram will help you to understand the concept of APM:

APM BLOCK DIAGRAM



The sources you have at your disposal are:

`No source:` It seems almost pointless explaining this one! It means no modulation source is routed!

`Modwheel:` This selects the modulation wheel as the control source.

`Bend:` This selects the pitchbend wheel or lever as the modulation source

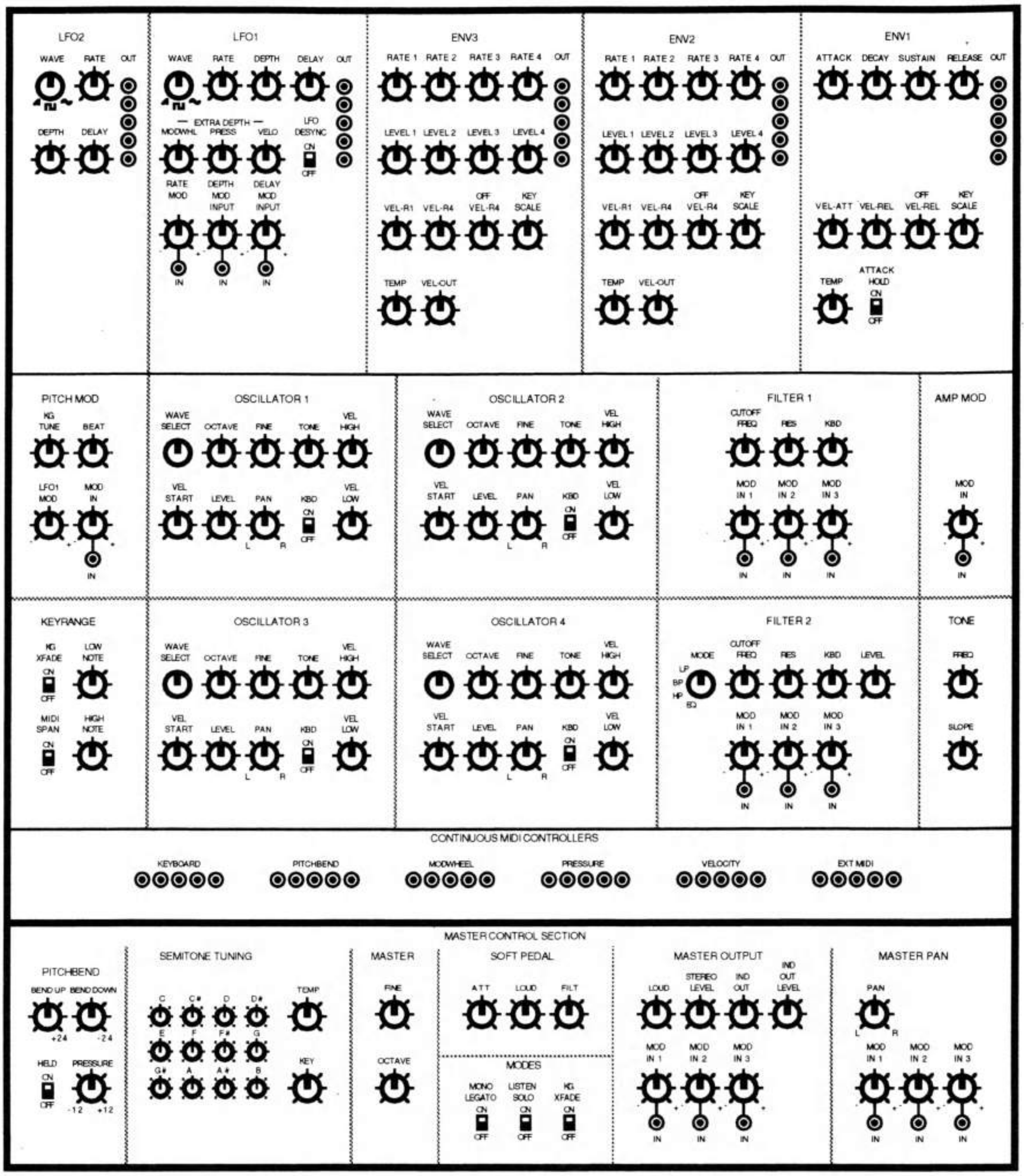
`Pressure:` This selects aftertouch as the control source. This is channel aftertouch and not polyphonic aftertouch.

External:	This selects the MIDI controller set in the external control field of the main MIDI MODE page. This can be BREATH (cntl#2), FOOT (cntl#04) or VOLUME (cntl#07). Using a MIDI merger on your keyboard would allow you to merge a breath controller with your keyboard and users of the Akai EWI MIDI wind synthesizer will no doubt be able to use this function to great effect when playing the S3200 directly from the EWI and selecting BREATH.
Velocity:	This selects velocity as the controller.
Key:	This selects keyboard position as the modulation control source.
Lfo1:	This selects LFO 1 as the modulation source. LFO1 may also be a modulation destination and it is possible to modulate its rate, depth and delay.
Lfo2:	No prizes for guessing that this selects LFO 2 as the modulation source.
Env1:	This selects the ADSR amplitude envelope, ENV1, as the modulation source.
Env2:	This selects the multi-stage ENV 2 as the controller.
Env3:	This selects the multi-stage ENV 3 as the controller.
! Modwheel	This selects the current position of the modwheel at the time of note-on as the current controller. Moving it whilst the note is held will have no effect - it is only its position at the point of note-on that has the effect.
! Bend	As above, this selects the current position of the bend wheel at note-on as the controller.
! External	This selects the current position of an external MIDI controller at the point of note-on as the modulation source. The choices you have for external MIDI control are breath (cntrl#2), footpedal (cntrl#4) and volume (cntrl#7). These are selected in the MIDI mode.

All of these are available to be routed in any amount to virtually any destination which include filter1 and 2 cutoff, LFO 1 rate, depth and delay, overall program amplitude and keygroup amplitude, pitch and pan position. To select them, you simply move the cursor to the modulation input field found on every module and scroll through the list.

As you can see, each keygroup has these modulation facilities separately available allowing an enormous amount of flexibility and this, combined with the all the other PROGRAM EDIT functions should keep you happy for a long time!

S3200 SYNTHESIZER FRONT PANEL



For those of you who are used to using synthesizers, you may like to visualize the EDIT PROGRAM functions as a synthesizer front panel as shown above. As you can see, it is quite impressive. The very topmost section are the controllers and include the two LFOs and the three envelope generators. The middle section comprises the pitch modulation inputs and the four sample zones (which may be likened to conventional oscillators) and the two filters, amplifier and tone control section. Below that are the continuous MIDI controllers and below that section are the master functions such as temperament and tuning, soft pedal functions and master output and panning.

The above panel is much like an old modular system in that the ASSIGNABLE PROGRAM MODULATION allows free patching of the devices. On an old modular

synth, you would physically connect the modules using patch cords - on the S3200, the equivalent is done in software.
And don't forget, you have thirty two of these synths in the S3200!

***NOTE**: Before you get on the 'phone to your Akai dealer or distributor ... we are not planning to make the above as a remote control panel for the S3200!! It would simply cost an absolute fortune!!*

NOTES ABOUT ASSIGNABLE PROGRAM MODULATION

1. The modulation method used in the S3200 is not complicated. Whereas before on our samplers all modulation was fixed, it is now assignable. In the TEST PROGRAM (the default program you get when you turn the instrument on), all the defaults have been sensibly chosen so that, for most sounds, when programming from scratch, you need not worry too much (the defaults are, in fact, mostly the same as those that were the fixed assignments of the S1000 and S1100).

When loading S1000 or S1100 library disks, the S3200 loads the assignments of the S1000/S1100 - i.e. the fixed assignments. Again, as a result, you need not worry about having to set these assignments yourself. On new library disks developed for the S3200, you will see the assignments made by our sound programmers. Please study these and see if you can learn from them.

2. It is possible to route the same controller twice (or three times in some cases) to the same destination. This is not an oversight but simply a way to keep things simple and open ended. If you were, for example to route LFO1 to filter cutoff three times at a value of +50 you would simply get three times more LFO sweep.

3. As just mentioned, you can route the same controller to the same destination several times. Please be aware that if you assign, for example, LFO2 to filter cutoff twice and set a value of +50 and -50 respectively, you will get no effect as the two cancel each other out.

4. You will note that assigning a particular controller to a destination WILL ROUTE THAT CONTROLLER TO ALL KEYGROUPS. The control inputs at any destination are not keygroup specific but affect all keygroups the same.

At first, the modulation system may seem a little difficult and maybe even confusing. Don't worry - if you are not into programming you can largely forget about it and just use the defaults. Similarly, if you do not like synthesizers and prefer instead to use the S3200 for the reproduction of high quality samples of acoustic instruments again, you need not concern yourself with these functions. If, however, you are one those who likes to experiment with sound and tweak those knobs, we feel sure you will appreciate the flexibility of this method.

CREATING AND EDITING A PROGRAM

In the S3200, we always use another program as the basis for a new one. There are several ways you can work this.

You can use an existing program from your sound library that closely resembles the one you wish to create. In the main PROGRAM EDIT page, copy this to a new program. This may be edited accordingly with new sample(s) assigned, envelopes changed, filter cutoff altered, etc..

Alternatively, let us say you have just taken five samples - you can create a program from scratch using the default TEST PROGRAM. Using this single keygroup program you could work three ways - you could just have the one keygroup and set that up with one of the samples and then, when you are happy with that, copy that keygroup four times and set the appropriate key span. Each individual keygroup may then be refined according to the sample assigned to it. Alternatively, you could simply copy keygroup 1 four times and, by selecting ALL, edit them together. The third method is where you copy keygroup 1 four times and work on each keygroup separately.

Our sound library programmers have several ways in which they work and the fact that you can combine all methods makes the S3200 very quick and convenient.

NAMING PROGRAMS - COPYING AND RENAMING

If you have already recorded your own samples, then this procedure should be familiar as it follows the same conventions.

To copy or rename a program, press the NAME key - this turns the front panel keys into letter entry keys and you may type in a name of up to 12 characters (upper case only). The +/< and -/> keys on the numeric keypad may be used to input backspace and spaces respectively. When naming, you will see this prompt:

```
LETTERS . . (NAME for numbers ENT to exit)
```

Pressing the NAME key again switches the numeric keypad from letters to numbers and you will receive this prompt:

```
LETTERS . . (NAME for letters ENT to exit)
```

You may press NAME again to access the numeric keypad's letters. When in the 'numbers' mode, the +/< and -/> keys input '+' and "-' to a name. Pressing NAME again reverts you to entering letters from the numeric keypad.

Alternatively, in conjunction with the the CURSOR keys which can be used to move the cursor around within the name, you can use the DATA control to scroll through characters.

When you have entered your name, press ENT and you will get this prompt:

```
                    Select: COPY REN exit
```

Pressing COPY will copy the original program - use this to create a new program.

If the program name is an existing one, the boxed area in the bottom right of the screen will show:

```
name: TEST PROGRAM
   *existing Prog*
```

and you will receive the following prompt:

```
!! MUST USE A DIFFERENT NAME !!
```

You must enter a unique new name.

Pressing REN will simply rename the currently selected program with the name just entered. If the name exists, you will be prompted as above and you must re-enter a unique name.

Pressing exit will exit the naming process altogether with no action taking place.

DELETING PROGRAMS

It is possible to delete programs using the DEL key - F8. Pressing this will give you the following prompt:

```
delete one program?                    GO ABORT
```

and you should press F7 or F8 accordingly. If you press GO, you will receive the prompt:

```
delete   3 released samples?    NO   YES
```

This is asking if you want to delete the samples contained within that program as well. If the samples are used in other programs, then you will not receive this prompt. If you wish to lose the samples, press F7 - YES - but if you need to keep them, press F8 - NO.

> ***NOTE***: *Deleting samples and programs is ultimately destructive. Please ensure that you have saved them to disk before deleting in case you want to come back to them at a later date.*

MAIN PROGRAM EDIT PAGE

Pressing EDIT PROG/K will display this screen:

```
PROGRAM EDIT    program: TEST PROGRAM  0%
    keygroups: 1        progs in mem:  1
      samples: 1         listen solo: ON
 KG crossfade: OFF   name: TEST PROGRAM
  Mono Legato: OFF      *existing Prog*
MAIN KGRP MOD MIDI OUT PAN TUNE DEL
```

This is the MAIN PROGRAM EDIT page. Here you have access to individual keygroup parameters, modulation functions as well as MIDI, output assignment and tuning via the soft keys. The fields on this page are:

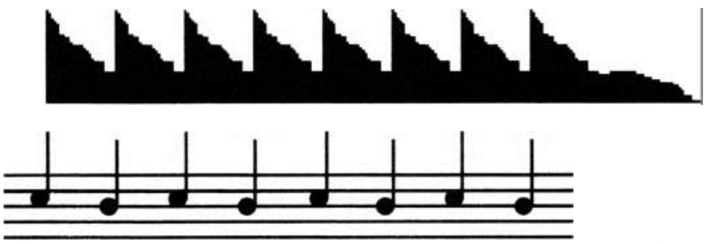
program: This shows the selected program name. Different programs may be selected for editing by scrolling with the DATA control. You may also select them using MIDI program change.

keygroups: This field is not accessible but shows the number of keygroups used in the selected program.

samples: This field is also not accessible and shows the number of samples used in the selected program. Please note that, because each keygroup has four zones, it is possible for there to be more samples in a program than there are keygroups. For example, a stereo program with 5 keygroups would typically show 10 samples (5 x L and R).

KG crossfade: This allows you to set crossfades for keygroups that overlap. As described above, this can be used to smooth out abrupt transitions between keygroups. You will note that this is a 'global' parameter that affects the whole program and so any keygroups that overlap will be subject to crossfading.

Mono legato: This special function turns the program into a monophonic program with single triggering. The effect of single triggering is that if you hold one note and play another, the pitch changes to the new note but the attack of the new note is not re-triggered. For example:



When Mono legato: is OFF, the attack for each new note is re-triggered.

When Mono legato: is ON, when the first note is played you hear the attack but if that note is held when new ones are played, only the pitch changes without re-triggering the attack.

Even when Mono legato: is ON, playing each note separately will cause the attack of each note to be re-triggered.

This function is extremely useful for emulating the playing styles and phrasing of solo instruments such as flutes, oboe, clarinet, saxophone, etc.. It can also be effectively used on solo and ensemble strings and brass. It is almost essential when playing synth bass sounds as it emulates the classic monophonic synth keyboard. You will find it useful too, when playing leadlines of any kind. Owners of the Akai EWI wind synthesizer will find this function invaluable in obtaining better expression and phrasing.

IMPORTANT NOTE REGARDING MONO LEGATO ON/OFF

NOTE 1: Because this function is playing a single sample (or group of samples when playing stereo or layered sounds) through the sustain or loop, if you were, for example, to do a long legato run from the top of the keyboard to the bottom, you would be transposing the original sample you started with several octaves down. For example, in a 7 keygroup flute program, if you were to start your run on C5 and finish at C2, the sound played on the last note would, in fact, be the C5 sample transposed 3 octaves down - the samples in the other keygroups would not be re-triggered as you cross the keygroup boundaries. As a result, if you sustained that last note, it may sound very strange indeed! Similarly, if you 'trilled' between C2 and C5, it would also sound quite odd (but then, so would a trill between C2 and C5!). This is brought to your attention so that you may make allowances when playing legato across keygroups. If you bear this in mind, you will find this function very useful and expressive.

progs in mem: This field is not accessible and shows you how many other programs are in memory at the time.

listen solo: This field allows you to select whether you wish to listen to other programs that have the same program number as you edit the currently selected program. The selection is ON or OFF. When listen solo: is switched to ON, you will only hear the currently selected program and when it is OFF, you will be able to monitor any other programs that share the same number.

This function is useful if you wish to edit one program whilst monitoring other programs in a multi-timbral set up. For example, you may want to edit the attack of a string sound you are sequencing - with listen solo: set to OFF, you would be able to adjust this as

the sequence is playing so that the attack time may be adjusted so as to make the string part 'sit' right in the track. You may also use this function to edit one layer of a layered set of programs.

No other fields are available on this screen.

The soft keys along the bottom of the PROGRAM EDIT main screen are:

MAIN	This shows the currently selected MAIN PROGRAM EDIT page.
KGRP	This takes you to the parameters concerned with individual keygroups. These include the keyboard spanning, filters, envelopes, sample assignments, pitch and level adjustments, individual output assignment, etc., for individual keygroups.
MOD	This takes you to the pages where you may set program modulation parameters such as pitch bend ranges, LFO 1 and LFO 2 parameters and parameters for the sostenuto pedal functions.
MIDI	This takes you to the MIDI page where you may set the programs MIDI channel, polyphony, transpose and other functions.
OUT	This takes to the OUTPUT pages where you may set the programs overall level, individual output assignment, effects send level, pan position and other parameters.
PAN	This takes you to the AUTO PAN section where you may set parameters that affect the program's auto pan functions.
TUNE	This takes you to the TUNE pages where you may set the program's overall tuning as well as select and create alternative tunings and temperaments.
DEL	This is not a page key but an 'action' key for deleting programs. Its function is described above.

We will now look at the functions of these pages. We will skip KGRP and MOD for the moment and look at the function of MIDI, OUT, PAN and TUNE first as these affect the program as a whole.

MIDI PAGE

Pressing MIDI displays this screen:

```
MIDI RESPONSE (PROGRAM) TEST PROGRAM   0%
program number:  1           PLAY-RANGE
  MIDI channel:  1            low  high
     polyphony: 32            C_0   G_8
      priority: NORM
  reassignment: OLDEST    transpose: +00
MAIN KGRP MOD MIDI OUT PAN TUNE
```

The top line of this page contains a field which allows you to change the program currently being edited. You may select different programs for editing here if you wish.

The parameters on this page are as follows:

`program number:` This field allows you to set the program number of the program. This is the number which will be called up on receipt of a MIDI Program Change message and corresponds to a patch number on a synthesizer. Unlike a synthesizer, though, the S3200 allows different programs to share program numbers, so when a Program Change message is received, all programs with the corresponding number will be selected simultaneously.

`MIDI channel:` It may come as no surprise to you to know that this parameter selects the program's MIDI channel! The choice is `OM` (omni) and `1` to `16`.

NOTE*: The MIDI channel used for the reception of MIDI program change commands is set in the MIDI mode. Program change messages can be received on another MIDI channel so that program selection can be made independently for the MIDI channel set here - please refer to the MIDI MODE section for more information on the program select channel.*

`polyphony:` This allows you to select how many notes (1 to 32) can be played at a time by this particular program. This should normally be left at the default, 32. If the program allows a large number of notes to be played, in a multi-timbral setup, you may find notes are 'stolen' from this program. Use this function (in conjunction with the `priority:` function described below) to prevent this.

The most obvious (and useful) function of this field is to create programs for hi-hats. With hi-hats, you want a closed hi-hat to shut off an open or half open hi-hat that may be sounding. By creating a program specifically for the hi-hats and setting the polyphony of that program to 1, you may achieve this. You should then give the hi-hats program the same number as the drums program they are presumably associated with. The same could be done with other percussion sounds such as open and closed triangles, guira, etc..

NOTE*: The S3200 is capable of playing up to thirty two notes (or 'voices' or 'samples' - call them what you will) at one time. If a keygroup is set to use four samples which will all be played when one key is pressed (i.e. four zones or keygroups layered together), then only eight notes can be played simultaneously. If it only uses one sample, then 32 notes can sound at one time. Please note that velocity and keygroup crossfading may use two samples simultaneously from one key, which will reduce the polyphony of the program.*

priority: This allows you to specify how notes will be 'stolen' by other programs if this is necessary. There are four settings: LOW, NORM, HIGH and HOLD. If a program is set to LOW priority, then notes from this program will be stolen first. If set to HIGH, then notes from other programs with lower priority will be stolen before they are stolen from this program. NORM is, of course, normal priority and, if any note stealing has to take place, this program will be affected no more or less than others with NORM assignment.

If you are playing a complex piece of music using many programs, it is a good idea to set important, lead-line sounds to HIGH, and less important background programs to LOW.

HOLD is a special priority. If a program's priority is set to HOLD, notes from this program can only be stolen by the same program.

re-assignment: The notes which will be stolen are determined by this parameter - either the OLDEST note will stop playing when a note is stolen or the QUIETEST one. Hopefully, with the 32 voice polyphony of the S3200, no voices need to be stolen unless you are driving it particularly hard.

PLAY-RANGE This allows you to set the overall keyboard range of the program and this will override any keygroup range settings made in the keygroup SPAN page. For example, even though your program's keygroups may extend up to G8, if you set, say, C4 as the high extreme in this field, no sound will be heard above C4. You may use this function to create keyboard splits with other programs of the same program number.

transpose: This sets the pitch of the whole program. You will note, however, that this is not a tune function as such but a MIDI pitch offset. For example, with this set to +12 and playing C3, you would not be playing the sample(s) at C3 an octave higher, you would actually be playing the samples that are assigned to C4.

Remember that pressing the MIDI soft key will re-display this page, toggling between MIDI note numbers and note names.

OUTPUT LEVELS PAGE

Pressing the OUT button takes you to the OUTPUT LEVELS page where you can control the audio output of the program from the S3200. You will receive this screen display:

```
OUTPUT LEVELS (PROGRAM) TEST PROGRAM  0%
     loudness:  80   LOUDNESS MODULATION
indiv output: OFF   velocity> loud: +20
 indiv level:  50        Key> loud: +00
stereo level:  99   Pressure> loud: +00
   stereo pan: MID
MAIN KGRP MOD MIDI OUT PAN TUNE
```

At the top right of the screen is the currently selected program name - this may be changed and another selected for editing by scrolling with the DATA control.

The parameters are as follows:

`loudness:` Here you may set the overall level for the program (00-99). Using this parameter, you may balance the program relative to others, especially useful in multi-timbral or layered setups. You may also use this so that, as you select different single programs, the levels are consistent. Of course, live on stage, you may prefer to use this parameter to boost a particular programs level when you take a solo.

The default for this parameter is 80. This offers the optimum range for velocity and other dynamics. Setting it higher than this will, of course, turn the overall level up but it will reduce the amount of overhead for dynamics. Reducing this parameter will give greater extremes of velocity range although you will note that will not be using the full resolution of the sampler.

You will note that this parameter also affects the level of the signal appearing at the individual outputs (see below) and at the real-time digital audio output .

`indiv output:` This allows you to select which of the 8 individual outputs the whole program will appear at. It also allows you to send the program to the S3200's internal effects. The default is OFF and you may also select 1 -8 , RVB (reverb), FX and R+F (reverb and effects simultaneously). You will note that the individual outputs are polyphonic and can use the full 32-voice polyphony of the S3200.

> ***NOTE**: This parameter works in conjunction with a field found in SMP2 (see below) where it is possible to route single keygroups to the individual outputs*

`indiv level:` This sets the level of the signal appearing at the output selected above. If OFF is selected, then this control has no effect and if REB, FX or R+F is selected, this

control acts as an effects send level to the reverb or effects section.

stereo level: This sets the level of the program as it appears at the L/R stereo outputs. Adjusting this has no effect on the level of the signal appearing at the individual outputs or the real-time digital outputs.

By setting this field to 00, you may use this parameter to mix a program out of the L/R mix completely if you are sending it to an individual output (see above). In this way, you may have some programs appearing only at the L/R outputs with other programs appearing only at the individual outputs.

stereo pan: This sets the overall pan position of the program. You will note that this may be affected by other pan settings elsewhere in the program such as when the auto pan facilities are used or when individual keygroups are panned.

***NOTE**: The above parameters of this page are duplicated in the MIX page of SELECT PROG. Any changes made there will be reflected here and vice versa.*

LOUDNESS MODULATION

The next set of parameters allow you to modulate the overall loudness of the program. This is our first encounter of the ASSIGNABLE PROGRAM MODULATION facilities.

There are three loudness modulation inputs and each one has a default controller selected. If you wish, these need not be changed - it is only in more specialised applications that you may wish to set something other than these defaults. The defaults are:

velocity > loud: This is a fixed assignment and sets how much velocity will affect the overall loudness of the program. The default is +20 which gives a sensible dynamic range although this may be changed if you wish. A setting of +50 will give you a very wide dynamic range where soft key presses will produce virtually no sound and hard key presses will output a very loud sound. A setting of -50 will give the converse effect - a hard key press will produce virtually nothing whilst a soft key press will give a loud output. At first, this may seem a bit strange to allow this but this does enable you to crossfade between programs using velocity - i.e. set one program to +50 and the other to -50.

You cannot select any other modulation source in this field - this is one of only two fixed assignments in the APM system.

key > loud: This sets how much key position will affect the overall loudness of the program. When set to positive value, the sound will be louder in the upper reaches of the keyboard and setting it to a negative value, the sound will be louder on lower notes. You may use this

function to balance the program level across the keyboard.

You may change the default modulation source from Key to any you like simply by placing the cursor where it says Key: and scrolling through the modulation options.

Pressure > loud: This sets how much pressure or aftertouch will affect loudness after a note has been played. Positive values will cause the sound to get louder as you press harder on the keyboard and negative values will, of course, have the opposite effect. You may use this function for expressive phrasing of such instruments as strings, vocal, wind and other such instruments to great effect.

By layering two programs and setting opposite values (i.e. + 50 on one program and - 50 on the other), you may use this function to crossfade between them using pressure. As an example, you could layer a distorted guitar program and a distorted feedback program and introduce the feedback element of the sound using pressure to create a powerful heavy metal guitar.

You can, of course, change the default selection from Pressure to anything you like simply by placing the cursor where it says Pressure: and scrolling through the modulation options.

The effect the loudness modulation parameters have on the overall loudness of the program depends on the modulation source you select. Here are some suggestions:

LFO 1 or 2 This will impart a tremolo effect on the sound and so is useful for simulating old rock and roll guitars where this effect was common in amplifiers. This may also be useful in simulating the tremolo effect found on organs. It may also be used to simulate the tremolo effect of woodwind on sustained notes. Using a triangle wave for modulation, it is particularly good for vibes sounds, especially those with a static loop. It is probably not suitable to simulate tremolando strings, however, using LFO modulation. Of course, it may be used for special effects.

Modwheel Use this, perhaps, instead of pressure.

Bend Use this instead of pressure or modwheel.

External Depending on the selection made in the MIDI MODE's main page, you could apply footpedal, volume or breath control to control the program's overall loudness. The breath option will be very popular with owners of the Akai EWI MIDI wind controller.

These three modulation input sources may be combined and mixed together. When layering programs, don't forget that identical mod sources in other programs that share the same number may be inverted for crossfade effects.

***NOTE 1**: You will note that if the* `loudness:` *parameter is set to full (i.e. 99) the output level of the S3200 is at maximum and so you will not hear any effect if one or more modulation sources are applied.*

***NOTE 2**: In some cases it is possible to overload the S3200 and cause distortion. This will normally only happen with particularly loud samples recorded at full level when excessive modulation is applied. Increasing the filter resonance may also lead to distortion in some cases. If this happens, back off the* `loudness:` *parameter.*

PAN PAGE

In this page you may set the characteristics of the auto panning functions. Pressing PAN will display this screen:

```
PAN (PROGRAM)                  TEST PROGRAM  0%
                              PAN MODULATION
      loudness: 80            Lfo2 > pan: +00
 stereo level: 99              Key > pan: +00
   stereo pan: MID        Modwheel > pan: +00

MAIN KGRP MOD MIDI OUT  PAN TUNE
```

As usual, the program name of the program currently selected for editing is shown here which you may change if you like.

The parameters are:

`loudness:`	This is a duplication of the `loudness:` parameter seen in the OUTPUT LEVELS page and is placed here for convenience to save you switching pages.
`stereo level:`	This is a duplication of the `stereo level:` parameter described in the OUTPUT LEVELS page and is placed here for your convenience.
`stereo pan:`	This is a duplication of the `stereo pan:` parameter and is also here for convenience.

`PAN MODULATION`

Again, we have three modulation inputs which can control panning. The defaults for these three control inputs are:

`Key`	This selects that key position will affect overall loudness. With positive setting (i.e. +50) the sound will pan from left to right across the keyboard and, if set to a negative value (i.e. -50), will pan the sound from right to left across the keyboard. You may use this parameter to create pseudo stereo samples out of mono ones. For example, with samples such as piano or marimba or vibes, you could create the stereo effect of microphones being placed at either end of their keyboards to produce a panning effect.
`Lfo2`	This will produce the classic auto panner effect with the sound gradually moving between left and right at a rate set by LFO 2. You can use this for special effects, of course, but one popular application is to use it to simulate a rotary speaker effect.

***USEFUL TIP**: When layering two identical samples with detune and hard left/right panning (see later in SMP1 - 3), using LFO 2 will cause each sample to swirl backwards and forwards in opposition - i.e. as one pans left, so the other pans right. You can use this to great effect to create very rich textures. By not setting too high a depth in this field, the effect can be quite subtle and produce a lot of movement. Of course, the other LFO may also be used for the same purposes.*

`modwheel:` This allows you to control pan position using the modulation wheel. This could be put to good effect in a solo line, perhaps, where, every time you introduce vibrato via the wheel, the sound pans around the stereo image.

***NOTE**; Unfortunately, due to limitations with the panning hardware, whilst slow sweeps work well, fast sweeps may, on some sounds, introduce some 'zipper noise'. Please be aware of this when setting pan modulation.*

As with all control inputs on the S3200, any combination of controllers can be mixed together. Here are a few suggestions:

`Bend:` Use this instead of `modwheel` perhaps.

`Pressure:` Use this instead of `modwheel` or `bend`.

`External:` Use maybe a footpedal to pan the sound around. EWI players may use breath control.

`Velocity:` You can use your keyboard dynamics to pan the sound around with loud sounds appearing at one output and soft sounds at the other.

`LFO1:` Use this as an alternative to `LFO2`. This LFO's extra facilities allow some very odd things to be done. Try applying this and modulating LFO1's rate with LFO so that the pan from side to side gradually speeds up and down or modulate LFO1's rate with the modwheel to emulate the slowing down and speeding up of a rotary speaker in an organ program.

`ENV1/ENV2/ENV3:` These may be put to good effect to pan the sound around according to the envelope of the sound. Perhaps ENV2 and ENV3 are the most interesting with their multiple rates and levels.

`! Modwheel:` Use this (and `! Bend:` or `! External:`) to re-position each new note according to the position of these controller.

And don't forget that layering samples in zones and panning them to extreme hard left and right will cause the two samples to crossover in the stereo image when these effects are applied. Layering two programs and setting the modulation amounts to opposite values can also yield some interesting panning effects.

THE TUNE PAGE

The next soft key is the TUNE key and this takes us, not surprisingly, to the main program tuning page. Pressing TUNE will give you this display:

```
PROGRAM TUNE (PROGRAM) TEST PROGRAM  0%
  C.  C# D.  D# E.  F.  F# G.  G# A.  A# B.
  +00+00+00+00+00+00+00+00+00+00+00+00
        Program tune: +00.00
     Tuning template:  EVEN
                 key:  C
MAIN KGRP MOD MIDI OUT PAN TUNE
```

As usual, the program name is displayed at the top right of the screen - a different one may be selected if you wish.

In this page, you set up different tuning temperaments for each program, if desired. If you are playing a percussive sample (for example, congas) in one program, which you do not want to correspond to standard Western chromatic equal temperament tuning, this is where you can alter things. Select the program whose temperament is to be altered on the top line of this screen. Use the CURSOR keys to select the note on the keyboard octave which will be retuned, and use the DATA control to alter the tuning away from equal temperament by ±25 cents (one quarter-tone). If you are retuning the C# key, for example, all notes played with the C# keys on the keyboard will be detuned by the amount you have set. You may tune the scale to anything you want which can be very useful for enriching orchestral sounds and also for setting your own special scales

To help you with this, there are also some alternative tuning templates which are selected in the field `Tuning template:`. These offer preset tunings which you can apply to the program.

The other parameter in this page is the `Program tune:` function. This transposes the program +/-50 semitones. This may be adjusted in very fine steps (100ths of a tone) for fine tuning the program.

Those, then, are the master pages for the program where you can set parameters that affect the program as a whole. In any of the pages described, you may always directly access another from the soft keys.

MODULATION PAGES

The next key we'll look at also affects the program as a whole but also has a direct influence on individual keygroups. These are the modulation pages where you may set the parameters for the two low frequency oscillators and the pitch bend. You may also set the parameters for the sostenuto pedal. These modulation sources may be applied to the filter, loudness, pitch and panning for a wide range of the usual modulation effects such as vibrato or for more outrageous synthesizer effects. These pages are accessed by pressing the MOD key.

Pressing the MOD key displays this screen:

```
PITCH (PROGRAM)                TEST PROGRAM  0%
PITCH-BEND
Bendwheel up:   2
Bendwheel dn:   2
     Pressure: +00
    Bend mode: NORMAL
MAIN BEND LFO1 LFO2 SOFT
```

The first page we encounter is the PITCH page where you may set the parameters associated with pitch bend. As usual, you may select a program for editing at the top right hand of the corner.

The pitch bend on the S3200 is no ordinary pitch bend that simply goes up or down as you move the wheel or lever. On the S3200 it is possible to set a different range for bend up and down as well as use pressure and a special mode is also available to make it more flexible. The parameters are:

Bendwheel up: This sets the range for bending pitch up with the pitchbend wheel or lever. The range is 0-24 semitones. The default is 2 semitones.

Bendwheel dn: This sets the range for bending pitch down and, again, the range is 0-24 semitones. The default is 2 semitones.

Pressure: As well as using the pitchbend wheel or lever, you may also use pressure to bend notes. The range is -12 to +12. You may only bend up or down depending on the selection made - unlike the bend wheel/lever, you cannot pitchbend both ways.

Bend mode: This is a mode select option that allows you to choose whether the pitchbend will happen on all notes or only on held notes. This is particularly useful on sounds with long releases. The options are NORMAL and HELD.

Let us say, for example, that you have a sound that has a long release and you are performing a solo that uses a lot of pitchbend. Using NORMAL, when you pitchbend the note, all the notes currently in the release stage of their envelope will also bend. Sometimes this is what you want but there can be times when this spoils the effect you are trying to create.

By selecting the HELD mode of pitchbend, ONLY THE KEY(S) YOU ARE CURRENTLY HOLDING DOWN WILL BEND and all those notes you are not playing but which are fading through their release stage will remain unchanged. If you release your finger from the key with the pitchbend up (or down), as the note dies away, if you let the pitchbend return to zero, that last note's pitch will not change. If you release just one note of a chord with pitchbend up or down, if you let the wheel or lever settle at zero, only the notes you are holding will bend.

The new pitchbend options on the S3200 allow for some very interesting performance techniques. By setting the pitch to up: 2 and down: 12, with a heavy metal guitar sample, you can emulate string bending up and an octave 'whammy bar dive bomb' down. Many things are possible.

THE LOW FREQUENCY OSCILLATORS

Pressing LFO1 will give you this screen display:

```
LFO1 (PROGRAM)               TEST PROGRAM  0%
Waveform: TRIANGLE      LFO desync: OFF
      FIXED    VARIABLE      EXTRA DEPTH
speed: 50          key: +00 | modwheel: 30
depth: 00          key: +00 | pressure: 00
delay: 00          key: +00 | velocity: 00
MAIN BEND LFO1 LFO2 SOFT
```

This is the page used for setting up LFO1. Again, the program name is shown at the top right of the screen and other programs may be selected for editing if you wish. The parameters for LFO1 are as follows:

Waveform:

This allows you to select from three waveforms. They are:

TRIANGLE ( ) - This gives a rising and falling effect. At around a setting of 75, this would normally be used for vibrato but can be assigned anywhere you like and can be used for filter sweeps, panning, amplitude modulation and slow pitch sweeps. This is a 'bi-polar' modulation waveform that 'rotates' around the note you are playing to give a natural vibrato effect.

SAWTOOTH ( ) - This waveform is used mostly for special effects. It rises slowly and falls abruptly. This is a 'uni-polar' waveform that jumps between the held note and the modulation level set at the destination.

SQUARE ( ) - This gives stepped 'up and down' effect. When applied to pitch it can be set to give trills or large octave jumps. This is a 'uni-polar' waveform that jumps between the held note and the modulation level set at the destination.

LFO desync:

This selects whether the LFO's (all 32 of them!) are synchronised or not. With LFO desync: ON, all the LFO's are not synchronised and so give a rich texture to

ensemble sounds when being used for vibrato. When `LFO desync:` is set to off, all LFO's are in sync. This latter option is probably more suited to use with synthesizer effects. You will find that slower LFO speeds are possible with `LFO desync:` set to off.

`FIXED`

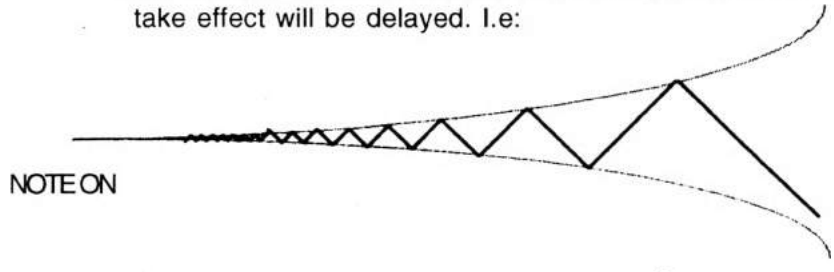
There are three fixed parameters for setting up LFO1. These are:

`speed:` This sets the rate of LFO1. This is variable between very slow for such effects as auto panning and filter sweeps and fast for vibrato and other effects.

`depth:` This sets the output level of LFO1. This acts as a master LFO modulation control for all destinations that have LFO1 routed to them. In this way, if you wish to change the modulation level going to *all* destinations easily, simply change this parameter.

***NOTE**: You may find that nothing happens when you route LFO1 to a destination and set a modulation level at the destination's input. This will be because LFO1's depth is set to 00. This may seem a strange default at first but the reason is to do with the use of the modwheel for vibrato. In order to be able to have the modwheel for vibrato as an immediate effect, the LFO depth needs to be at 00 otherwise, there will be constant LFO modulation of pitch. It is only when you are using LFO1 to apply a constant vibrato or for something other than vibrato in specialist applications that you will need to specifically set the depth control here.*

`delay:` This parameter sets the time it takes for the LFO1 effect to take place after a note-on. At 00, the effect will be instantaneous but at 99 the effect of LFO1 to take effect will be delayed. I.e:



`VARIABLE`

The three fields below this apply to the modulation inputs of LFO1. Although not specifically named, they relate to the parameters directly to their left - i.e. speed, depth and delay. The default for these parameters is key so that you may affect speed, depth and delay according to keyboard position, thereby emulating the fact that, for example, high violins' vibrato is often slightly faster, deeper and less delayed than low cellos or contrabasses. These parameters allow you to create quite rich orchestral textures where vibrato is never constant across the keyboard range.

Of course, you may select other modulation options for these inputs. Here are a few ideas:

Using LFO1 as an assignment to control its own rate will change the symmetry of the waveform and you can use this to create new modulation waveforms (i.e. using a square wave to modulate itself will create an asymmetric pulse wave). Using LFO1 to modulate its own output level does create an effect but this can best be described as subtle! Applying LFO1 to modulate its own delay is also very subtle to the point that, with the exception of very slow sweeps, you probably won't hear anything!

Applying a slow LFO2 to speed will give a gradual speed up and slow down of LFO1 rate. This can be used for special effects, especially in synth sounds - for example, apply a slow LFO1 mod to the filter cutoff and assign a slow LFO2 to control LFO1's speed, setting a value of + 50. Assigning it to depth will cause the effect of LFO1 to whatever destination it is applied to increase and decrease at a rate set by LFO2. Applying LFO2 to delay will only have an effect at the point of note-on.

Selecting ENV1 or ENV2 to control speed will give changes in speed according to the shape of the envelope. Applying it to depth allows you to 'shape' the output of LFO1. Applying it to delay will have no effect.

Selecting any of the continuous controllers will allow to change the parameters in real-time as you play. For example, routing modwheel to speed will let you change the modulation rate as you introduce vibrato (note that routing modwheel to control depth is not really worthwhile as this is a fixed assignment anyway in the EXTRA parameters described below). Assigning any of the continuous controllers to delay is not going to produce much effect except at the point of note on (pressure will have no effect on delay at all, by the way).

Any of the '! ' continuous controllers will only have an effect at the point of note on as well - you may use these to set speed, depth and delay in real-time as you play.

The EXTRA parameters allow you to introduce more modulation and these have fixed assignments which are:

modwheel: This allows you to set the level of modulation that will be introduced via the modwheel. This works in conjunction with the depth: parameter. Even with depth: set to 00, you may still use the modwheel for vibrato and other modulation. With depth: set at anything other than 00, this will set a basic level of modulation that will be present in the sound all the time and the modwheel will introduce more again. With the LFO depth set at 99, the modwheel will have no effect because the LFO output is now at maximum. The default for this parameter is 30 so that modwheel is instantly available for vibrato without you having to do anything.

pressure: As with modwheel: , this selects that pressure can be used to introduce modulation. The same principles apply that even with depth set to 00, you may still use pressure for vibrato and other modulation. With the depth set at anything other than 00, this will set a basic level of modulation and pressure will introduce more. With depth set at 99, pressure will have no effect because the LFO output is now at maximum.

velocity: This allows you to use note on velocity to introduce LFO1 modulation. With a positive value set here, playing a note hard will introduce a level of modulation which may be augmented using the modwheel or pressure or which can be modified by whatever mod source is assigned to the depth: parameter.

SETTING LFO1 MODULATION DEPTH

The master output of the LFO is set using the depth: control and this has to be set to something other than 00 for there to be any effect unless the modwheel is moved - you may assign the LFO to a destination and set that destinations modulation level to maximum only to find that there is no effect. The reason for this is that the depth: control in this page is not set or the modwheel is not up.

A useful function of this master depth control is that in the case where you have applied LFO1 to several destinations and you wish to increase or reduce the level of modulation going to all the destinations, you may change the modulation level to all destinations with just the one depth: control instead of having to individually adjust every destinations' modulation input level.

***NOTE**: Because it is felt that LFO1 will mostly be used for vibrato effects, the default in the PTCH page (see later) is set so that simply setting a value in the* depth: *field will apply vibrato. If you intend to use LFO1 for something other than vibrato (i.e. slow filter sweeps, panning, amplitude modulation, etc), then be sure to set the LFO1 parameter in the PTCH page to 00 unless you really want the sound of an American police siren!*

Once you have set the LFO1 parameters, you may return to the main program edit page by pressing MAIN

LFO2

Pressing the LFO2 key will take you to the second LFO page:

```
LFO2 (PROGRAM)                  TEST PROGRAM  0%
Waveform: TRIANGLE
   speed: 01
   depth: 99
   delay: 00

MAIN BEND LFO1 LFO2 SOFT
```

This is a simpler LFO for auxiliary modulation purposes. Whilst LFO1 is normally used for vibrato via the modwheel or pressure, LFO2 can be used for secondary modulation purposes such as filter sweeps, amplitude modulation, panning, etc..

As usual, the program name is displayed in the top right hand corner. The parameters are as follows:

Waveform: This selects the modulation waveform. The choices are:

TRIANGLE (∿) - This gives a rising and falling effect. At around a setting of 75, this can be used for vibrato but can be assigned anywhere you like and can be used for filter sweeps, panning, amplitude modulation and slow pitch sweeps. This is a 'bi-polar' modulation waveform that 'rotates' around the note you are playing to give a natural vibrato effect.

SAWTOOTH (◿) - This waveform is used mostly for special effects. It rises slowly and falls abruptly. This is a 'uni-polar' waveform that jumps between the held note and the modulation level set at the destination. It can be inverted at the modulation input stages of each destination to give downward sweeps.

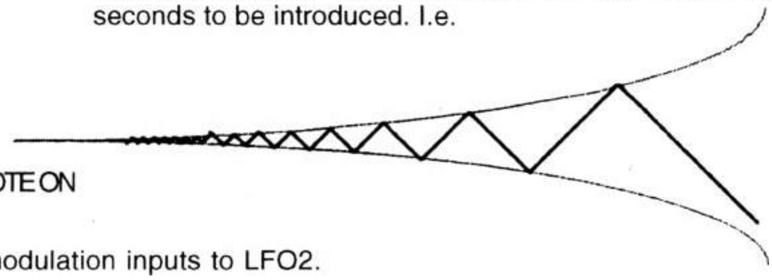
SQUARE (⊓⊔) - This gives stepped 'up and down' effect. When applied to pitch it can be set to give trills or large octave jumps. This is a 'uni-polar' waveform that jumps between the held note and the modulation level set at the destination. Like the sawtooth wave, it can be inverted at the input stages of each destination.

speed: This sets the rate of LFO1. This is variable between very slow for such effects as auto panning and filter sweeps and fast for vibrato and other effects. The default here is slow as it is assumed you will want to use this LFO for such things as filter sweeps and slow panning effects, etc..

depth: This sets the master output level for LFO2. Unlike LFO1, its default is 99 so you will instantly hear the effect of LFO2's modulation as soon as you apply it to any destination.

delay: This sets the delay between a note-on occurring and the effect being introduced. At 00 the effect will be

instantaneous and at 99 the effect will take some 5 or 6 seconds to be introduced. I.e.



There are no modulation inputs to LFO2.

You may use LFO2 for a number of things. As mentioned above, when LFO1 is tied up doing vibrato via the modwheel, LFO2 may be used to affect things such as panning, filter sweep and amplitude modulation. Of course, there is no reason why you shouldn't use LFO2 for vibrato either and mixing it with LFO1 as a source of vibrato can create some rich ensemble textures. Many interesting things are possible when modulating LFO1 with this LFO - at extreme settings you can make LFO1's modulation speed up and slow down for special sound effects or synth sounds but, if you're trying to breathe some life into some dead string samples, for example, you may like to use LFO2 to *slightly* modulate LFO1 thereby affecting the vibrato very subtly so as to eliminate the inherent 'cyclicness' of LFO modulation. No doubt you will find variations of your own.

SETTING UP THE SOFT PEDAL

Pressing SOFT will display this screen:

```
SOFT PEDAL (PROGRAM)        TEST PROGRAM  0%

  loudness reduction: 10
       attack stretch: 10
         filter close: 10

MAIN BEND LFO1 LFO2 SOFT
```

This final page in the modulation section allows you to set the response of the S3200 to the soft pedal (MIDI controller 67) or front panel footswitch. This can be very useful in obtaining better expression for piano sounds. The parameters are very simple and are as follows.

The `loudness reduction:` parameter determines how the volume of the sound will be affected when the pedal is pressed. The higher the number, the greater the amount of volume reduction.

The `attack stretch:` parameter allows you to soften the attack of the sound and affects the attack times of the envelope generators. Again, the higher the value, the greater the effect. For many acoustic instruments, especially string and woodwind sounds, when played quietly, their attack times also change slightly so this parameter can be put to good use.

The final parameter, `filter close:` , determines by how much the filter cutoff frequency will be reduced when the pedal is pressed thereby simulating the effect that acoustic instruments generally lose some upper harmonics when played quietly.

KEYGROUP PARAMETERS - CREATING KEYGROUPS

All the previous parameter descriptions have so far been concerned with global or master changes to the program - i.e. not keygroup specific. This next section delves deeper into PROGRAM EDIT and examines the individual keygroup parameters. These include keyboard mapping, sample assignment and, of course, the filters and envelope generators.

In the main PROGRAM EDIT screen, pressing KGRP will display this screen:

```
KEYGROUPS                        TEST PROGRAM  0%
  Keygroups in Program:  1  (+/-)
active keygroup number:  1
                  Span:  C_0  -  G_8

MAIN KGRP SPAN FILT ENV SMPL PTCH
```

This gives you access to the keygroup pages.

The primary function of this page is to create and copy keygroups although it is also possible to set key ranges (although this is possibly best done in the SPAN page - see later).

The parameter, `Keygroups in Program:`, shows you how many keygroups currently exist in the selected program. To copy keygroups, simply move the cursor to this field and press the +/< key on the numeric keypad as many times as you need keygroups - the amount you have copied will be shown in this field. To delete keygroups, simply press the -/> key on the numeric keypad.

If you already have some keygroups in the program and you specifically want to copy, say, keygroup 5, move the cursor to active `keygroup number:` field and select 5.

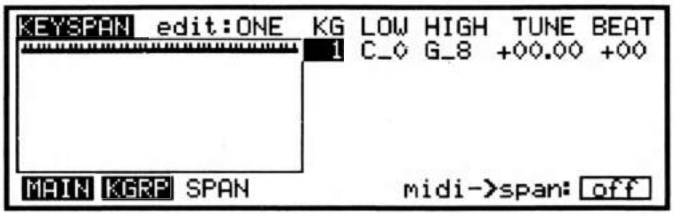
> ***TIP***: *A quick way of selecting keygroups is to hold the EDIT PROG mode select key and then play the appropriate note on the keyboard. For instance, in the above example, if KG5 was between B4 and F5, pressing any notes in that range whilst holding the EDIT PROG key will select that keygroup for you. As you play the key, the selected keygroup is shown as is its key span. This functions works everywhere in EDIT PROGRAM.*
>
> *You will note one exception to this, however. If LISTEN SOLO is switched to OFF so that you are monitoring other programs, this function does not work, the problem being that the S3200 cannot get information about which keygroup is being played because so many programs (and hence keygroups) are active at one time.*

You may also set the keygroups note range in the `Span:` field but, as mentioned above, you may prefer to use the graphically assisted SPAN page for that, described next.

Pressing the MAIN key will take you back to the main PROGRAM EDIT screen and give you access to the other global pages.

MAPPING OUT YOUR KEYGROUPS - SETTING KEYSPAN

Pressing SPAN will display this screen:



This is where you can set up the note ranges for the keygroup. You can see a graphic representation of the keyboard to the left of the screen. As you adjust the LOW and HIGH parameters for a keygroup, you will see its range depicted in the graphic representation of the keyboard to the screen's left. The notes may be represented by name or as note number simply by pressing the SPAN soft key again.

The above screen display shows a simple program with one keygroup in it. This next screen shows a typical program with several keygroups side by side.

The parameters are as follows:

edit: This toggles between ONE and ALL and allows you to choose between editing either one single keygroup or all keygroups simultaneously. You will find this function of every keygroup page and it can be invaluable in making up and editing programs quickly. In a complex program, you may select ALL to do all the basic work and then switch to ONE to individually fine tune the keygroups.

KG Below this field are the keygroups and their note ranges can be seen alongside them. You may move the cursor directly down this line using the cursor keys for quick access to a particular keygroup (don't forget the business of holding the EDIT PROG button whilst playing a note on the keyboard, either).

LOW HIGH Below these two fields are shown the lowest and highest notes for the keygroups. These may be set by moving the cursor to them and using the DATA control to input notes or, when displaying the notes numerically, you may type in a number from the numeric keypad. Another way to input notes is directly from the keyboard.

By setting the midi -> span: soft key to ON and placing the cursor on the low note of keygroup 1, as

you play the keyboard, so the notes will be entered. The cursor will jump to each low and high note in the list eventually 'wrapping round' to rest on KG1. This is a very fast way to set up keygroup note spans and a whole program can be set up in seconds!

NOTE: *If ALL is selected in the* edit: *field, changing a value in the low or high fields will affect ALL notes equally. Pay attention because you could seriously affect your programs keyspan with one slip. You will note that this does not apply when inputting notes from the keyboard and* midi -> span: *is on.*

TUNE — This allows you to tune the keygroup up or down in semitones and cents.

BEAT — This introduces a fixed tuning offset and can be used when layering samples to provide a chorus effect. Unlike the TUNE parameter, this offset is constant no matter what the played pitch of the sample is.

midi -> span: — This soft key switches on or off the facility to input notes from the keyboard. At first, with the function being so fast and convenient, it may seem a bit strange to want to disable it but, on the S1000 and S1100, we discovered that many people need to be able to change note ranges whilst receiving data from a sequencer that is playing. In this case, with this function switched on, the sequencer would completely re-program the key ranges! This on/off function should guard against that.

To exit this page and to access other pages, press KGRP - this will give you access to other keygroup functions - or press MAIN to return you to the main PROGRAM EDIT page and the master program functions.

ASSIGNING SAMPLES TO KEYGROUPS AND ZONES - SMP1

Before we look at the filters and envelopes, we need first to look at assigning samples into the keygroups. This is done in the SMPL page. Pressing SMPL gives this display:

```
C_0 - G_8 KG: 1 ED:ONE  TEST PROGRAM  0%
zn   sample      V-lo V-hi pitch
1 SINE              0  127 TRACK Xfd
2                 ? 0    0 TRACK ON
3                 ? 0    0 TRACK
4                 ? 0    0 TRACK      1234
MAIN KGRP SMP1 SMP2 SMP3
```

This is SMP1 (the first page to do with assigning samples - there are three in total). Here, for the first time, we catch a glimpse of the zones mentioned at the start of this section. In this example, the test program has one sample in it in zone 1. This has a velocity range of 0-127 and a key span of C0-G8 and so will play across the entire keyboard. Compare that with this screen which shows a typical three way velocity switch for a bass program:

```
C_0 - G_8 KG: 1 ED:ONE  SLAP BASS 1   0%
zn   sample      V-lo V-hi pitch
1 SOFT BASS         0   65 TRACK Xfd
2 HARD BASS        66   95 TRACK ON
3 PULL BASS        96  127 TRACK
4                 ? 0    0 TRACK      1234
MAIN KGRP SMP1 SMP2 SMP3
```

Here you should be able to get a feel for the concept of 'zones'. We have three separate samples SOFT BASS, HARD BASS and PULL BASS and their velocity ranges are split 0-65, 66-95 and 96-127. You will also note the nice graphic depiction alongside it for each of the three zones! Playing within those velocity ranges will play each sample accordingly allowing you to emulate the many tones available from just one note of a real bass guitar.

If the velocity ranges overlapped (i.e. 0-70, 63-100 and 93-127) Xfd (seen to the left of the graphic display) is switched to ON, then the velocity zones will crossfade giving a smoother response in some cases.

***NOTE**: If you play between two overlapping ranges (i.e. in the above example, if you play at a velocity of 96), you will actually be playing two voices out of the possible 32. With such generous polyphony, this shouldn't be a problem unless you are driving the S3200 particularly hard.*

Another way to use the zones is for stereo samples and for layering sounds. For example:

```
C_0 - G_1 KG: 1 ED:ONE  PIANO         0%
zn   sample      V-lo V-hi pitch
1 PIANO C1    -L    0  127 TRACK Xfd
2 PIANO C1    -R    0  127 TRACK ON
3                 ? 0    0 TRACK
4                 ? 0    0 TRACK      1234
MAIN KGRP SMP1 SMP2 SMP3
```

This shows the assignment for a stereo piano sample. The left and right samples are assigned to zones 1 and 2 respectively and both given a range of

0-127. These two zones would be panned hard left and hard right in SMP2 (see below) to give stereo reproduction through the L/R outputs.

***NOTE**: For stereo samples to play in stereo, they must be in the same keygroup and set up as shown above.*

For layering sounds, you might like to set something like the following:

```
C_0 - G_1 KG: 1 ED:ONE  FAT STRING    0%
zn   sample        V-lo V-hi pitch
1  MOOG STR C1       0  127  TRACK Xfd
2  MOOG STR C1       0  127  TRACK ON
3                 ?  0    0  TRACK
4                 ?  0    0  TRACK     1234
MAIN KGRP SMP1 SMP2 SMP3
```

Here we have assigned two identical synth samples to zones 1 and 2 and, as in the stereo program above, both have a velocity range of 0-127. In the SMP2 page, these can be detuned against each other and panned hard left and right to create a fat, warm, pseudo-stereo synth sound. This is a quick way of achieving this kind of sound. Of course, they don't have to be identical samples - anything will do. The trick with this method is that both samples share the same processing with the filters and envelope generators making adjustment and editing very simple.

If you want to be more adventurous, you could use totally separate keygroups for layering synth sound (or acoustic sounds, for that matter) and each could have different envelope and filter characteristics.

The parameters in this page are:

C_0 - G_1 — This shows the current keygroup's key range as set in the SPAN page. It can be altered here if you wish.

KG: — This shows the currently selected keygroup and allows you to select others using the DATA control. As always in PROGRAM EDIT, you may quickly select a keygroup by holding the EDIT PROG select key and playing an appropriate note on the keyboard.

ED: — Here you can select to edit one or all keygroups.

***NOTE**: Selecting ALL doesn't apply to assigning samples when using the DATA control. Only one sample is selected and the other keygroups remain unchanged even if ALL is selected.*

Of course, as usual, the name of the current program is shown at the top right hand of the screen.

zn — This shows the four zones in the column below. You will notice that in all SMP pages, as you play, a small dot appears alongside the zn field to signify which zone is playing - this is useful in identifying which sample is playing in a complex velocity switched program.

sample — This shows the sample(s) currently assigned to the zone(s). If a sample name is assigned but it does not exist in the S3200's memory, a '? ' will appear

alongside it to indicate that it is missing. To assign a sample, simply move the cursor to this field and scroll through the available samples in memory with the DATA control.

To delete a sample from a zone, simply move the cursor to it, press NAME and replace the name with blanks. To erase a group of samples in the same zone in different keygroups, do the same but with ALL selected.

USEFUL HINT FOR ASSIGNING SAMPLES! *Assuming you have a lot of samples to assign and you have made up a program with sufficient keygroups, go to keygroup 1 and press MARK/#. Now move the cursor to the sample assign field below and select the first sample. NOW PRESS JUMP/. (this will take you to the KG field again) and then select another keygroup. Now press JUMP again to toggle you back to the sample assign field and select your next sample, press jump, new keygroup, jump, new sample, etc.. Once you get some practice you can assign the samples to quite a complex program in no time at all. Please note, that if you record (or load) your samples in the order they are to be assigned, then the process is even quicker.*

V-lo — This sets the low velocity range for the zones.

V-hi — This sets the high velocity range for the zones.

Pitch — This allows you to select between TRACK and CONST. When TRACK is selected, then the sample can be played across the keyboard range as normal. When CONST is selected, then the sample(s) will play at a constant pitch of C3.

USEFUL TIP TO DRUM SAMPLISTS! *Sample all your drums on C3 in EDIT SAMPLE, assign them to any key you like in EDIT PROG and simply switch on CONST for all keygroups. They will now play back at exactly the pitch they were sampled at. In this way, you don't have to worry about setting notes when sampling and then trying to match them up in EDIT PROGRAM. Other non-pitched samples such as sound effects, breakbeats, drum loops, etc., can be treated the same way.*

As mentioned before, there is a small box to the right of the screen that displays graphically the status of the four zones' velocity ranges

SMP2

Once you have assigned your samples, you may go to the next sample page by pressing SMP2. Here you may tune and pan your samples. You will get this screen:

```
C_0 - G_8 KG: 1 ED:ONE    TEST PROGRAM  0%
zn  sem.cnt  loud  filt  pan  out  playback
1   +00.00   +00   +00   MID  OFF  AS SAMPL
2   +00.00   +00   +00   MID  OFF  AS SAMPL
3   +00.00   +00   +00   MID  OFF  AS SAMPL
4   +00.00   +00   +00   MID  OFF  AS SAMPL
MAIN KGRP SMP1 SMP2 SMP3
```

The top line of the screen is exactly as in SMP1 and displays the currently selected keygroup's note range and keygroup, whether you are editing one

individual keygroup or all of them as well as the program name. The other fields are:

zn	Again, this shows the zone number in the column below.
sem.cnt	This column allows you to tune each sample in each zone separately. You may tune the sample in semi tones and cents over a very wide range.
loud	You can fine tune the loudness for each zone in each keygroup to balance them up against each other. You will note that if the master loudness parameter in the OUT page of the main PROGRAM EDIT screen is set to 99, this control will only have effect downwards (i.e. when setting a -value).
filt	This parameter allows you to fine tune the filter cutoff slightly to maintain a consistent tone between keygroups.
pan	This allows you to pan each zone in each keygroup between the main L/R outputs.
out	This allows you to assign each zone in each keygroup to its own output for separate processing on an external mixing console if you wish. This has a relationship with indiv output: field found in the OUTPUT LEVELS page.

If the whole program is assigned to an individual output, that output will be shown here - i.e. if the program is set to appear at output 4, then 4 will be shown for every keygroup here. You may change this per keygroup so that whilst some sounds come out of 4, other keygroups appear at other outputs. This is particularly useful on drums where you often need to mix drums through an external mixer to add EQ, reverb, compression, etc..

Even if the main indiv output: field is set to OFF or RVB (reverb), FX or R+F (reverb and effects), you may route individual keygroups to separate outputs. This may be useful in a drum program where things like toms, cymbals and some percussion appear at the S3200's stereo outputs whilst important drums like bass drum and snare are assigned to appear at individual outputs.

You may also route individual keygroups to the S3200's internal effects by selecting RVB (reverb), FX or R+F (reverb and effects).

You will note that whatever the output assignment is here, the level is taken from the master OUTPUT LEVELS page and is set at the indiv level: field.

playback	This allows you to change the loop and playback characteristics of the sample. Normally, these are set in ED.2 of EDIT SAMPLE but they may be changed here

if you wish. This will not affect the 'raw' samples' loop and playback characteristics but can be used within the context of particular programs. This eliminates the need for copying the same sample several times (and hence wasting memory) to achieve the same effect. The options available to you are:

AS SAMPLE plays back the sample exactly as set up in the ED.2 page (loops included).

LP in R is the same as the LOOP IN RELEASE mode of the ED.2 page.

LP til R is the same as LOOP UNTIL RELEASE.

NO LOOPS is self-explanatory!

TO END is the same as the ED.2 PLAY TO SAMPLE END.

The ability to reset the playback parameters of a sample allows you a lot of flexibility - the same sample can be used in different ways in different programs.

SMP3

Pressing SMP3 takes you to the last of the three sample pages in PROGRAM EDIT. here you may set velocity start time for the sample(s) assigned to the currently selected keygroup. The screen display looks like this:

```
C_0 - G_8 KG: 1 ED:ONE   SLAP BASS 1   0%
zn  vel>start
1   +0000
2   +0000
3   +0000
4   +0000
MAIN KGRP SMP1 SMP2 SMP3
```

This page allows you to determine the way in which velocity affects the playback starting point for each sample in a keygroup. This figure is variable from +9999 to -9999. The higher the positive number, the earlier in the sample playback will start relative to the key velocity (i.e. a high key velocity will start playback earlier in the sample). A negative number has the opposite effect (a high key velocity will start playback later in the sample than a low key velocity). This effect is particularly useful for simulating percussion instruments (try it with a bass drum). It can also be very effective with such instruments as a heavily bowed cello - by setting a high positive value, hard keystrokes will play the aggressive bowing whilst soft keystrokes will not. The same could be done with overblown saxes or flutes. Something similar could be done with synth bass samples.

No other functions are available in this page.

In all of the sample pages, you may select between them via the three SMP soft keys. To return to the keygroup screen to access the other keygroup functions, press KGRP and to return to the main PROGRAM EDIT screen to access the 'global' program functions, press MAIN.

THE FILTERS

The S3200 is equipped with two banks of filters plus a simple EQ section. Filter 1 is a standard 12dB/octave lowpass filter with resonance whilst Filter 2 is a multi-mode filter offering lowpass, band pass and highpass filtering with resonance. Another mode is that of a simple EQ that offers one band of variable cut and boost and Q. This is not designed as an equaliser as such but is included for the creation of special effects. Finally, a very simple tone control is included that can cut or boost bass or HF and this may be used to remove unwanted noise from a sound. These will be described later. For the moment, let us have a look at the lowpass filters.

Pressing KGRP displays the main keygroup function select where you may access the last of the EDIT PROGRAM functions discussed here, the filters and the envelope generators:

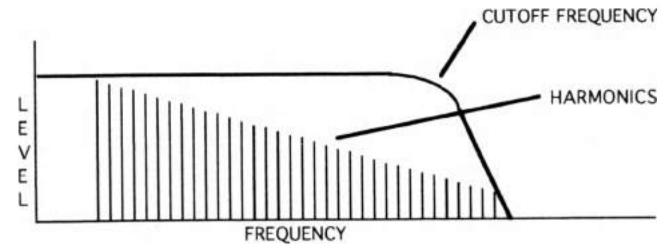
```
KEYGROUPS                          TEST PROGRAM  0%
  Keygroups in Program:  1  (+/-)
active keygroup number:  1
                  Span:  C_0 - G_8

MAIN KGRP SPAN FILT ENV SMPL PTCH
```

From this page press the FILT key to take you to the Filter 1 page:

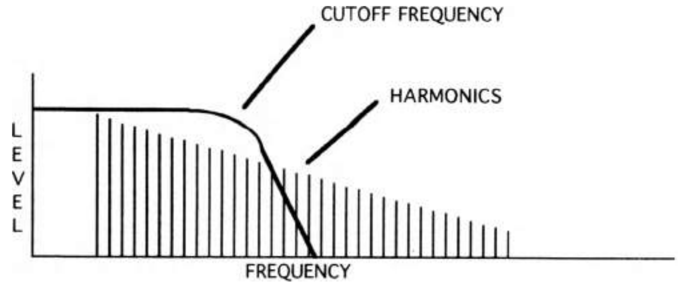
```
FILTER  KG: 1 ED:ONE               TEST PROGRAM  0%
C_0 - G_8
   frequency: 99     velocity > freq: +00
  key follow: +12        Lfo2 > freq: +00
   resonance:  0         Env2 > freq: +00

MAIN KGRP FLT1 FLT2 ENV1 ENV2 ENV3 TONE
```

As mentioned, Filter 1 offers 12dB/octave lowpass resonant filters as found on many analogue synthesizers. As well as using them for the tonal modification of acoustic samples, they also allow you to totally transform a sound.

'Lowpass' means that the filter will allow low frequencies to pass through unaffected whilst high frequencies are removed.

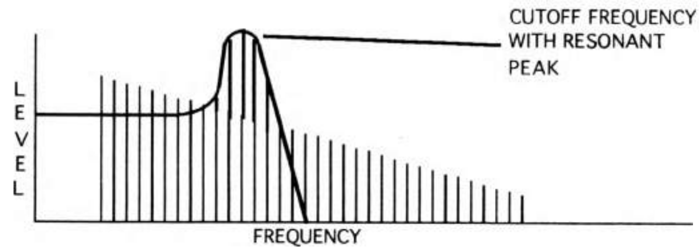


As the cutoff frequency is moved downwards, so high frequencies are gradually removed.



This is very convenient when dealing with samples of acoustic instruments because as notes die away, they tend to lose their higher frequency content first. By applying an envelope generator to the filter's cutoff frequency, we can emulate that on the sampler. Another property of sound is that when it is played loud (ff) it is generally brighter than when it is played softly (pp). Using the filter and applying velocity (or controlling the output of the filter's envelope generator using velocity) we can have a certain amount of control over tonal dynamics as well.

The S3200's filters are equipped with resonance. This allows you to selectively boost the area around the cutoff frequency thereby boosting certain harmonics.



Although it has some uses when trying to accurately reproduce some acoustic samples, its use is more suited to synthesizer effects. You will note, however, that when resonance is used, because of the gain increase that takes place, it can be easy to overload the output stages of the S3200, especially with certain sounds. As a result, take care to watch levels. The distortion produced by digital circuitry is not as pleasant as that produced by the old synths (unfortunately!!) so we can't expect the pleasant overdriven sound from a sampler - if you are after that sound, then sample the distortion from the synth!

Let's now have a look at the filter's parameters.

The parameters across the top of the page follow the usual convention and allow you to select the keygroup for editing, select whether you wish to edit just one keygroup or all keygroups simultaneously and, of course, you can select another program for editing if you wish. The other fields on this page are:

`C_0 - G_8`	This shows the current keygroups key range.
`frequency:`	This allows you to set the filters cutoff frequency. As you decrease this from 99, you will remove the upper harmonics resulting in a softer tone. This can be used to great effect on acoustic instruments (especially those that have ben looped) with velocity and envelope shaping to restore the natural harmonic dynamics and movement to the sound. On synths, you may sweep this

with all sorts of controllers for a wide range of synth sounds.

key follow: Here you may set the keyboard to track the filter. This is so that you can achieve an even tone across the keyboard range. +12 is the default and this tracks the filter octave for octave - i.e. for every shift of pitch of one octave, there is an according shift in harmonics.

resonance: This allows you to sharpen the point at the cutoff frequency thereby emphasizing the harmonics at that point. The sound changes from a soft 'waaa' effect to the characteristic 'weeow' effect with high resonance settings. The range is 0-15. High resonance settings can be used for classic synth bass sounds and, simply by sampling raw synth waveforms (i.e. without using the synths filter, etc.), these may be used as the basis for some powerful synth sounds through these filters.

NOTE: *When increasing the resonance, depending on the sound, some very loud peaks may be created as certain strong harmonics get boosted. This can result in distortion. To reduce any distortion you may have, reduce the loudness control in the OUT pages.*

NOTES ABOUT THE FILTERS

S1000 and S1100 owners may sometimes notice a slight difference in sound quality when comparing between those samplers and the S3200. This is because the S3200's filters are totally different. Whereas the S1000/1100 used 18dB/octave filter, the S3200 uses 12dB/octave filters. The effect of this is that the S3200's filters let through a few more harmonics than the S1000 and S1100 and so, some sounds may sound different.

Also, if any S1000/S1100 sounds use the filter dynamically, then the effect is likely to a little bit different when played back on the S3200.

This is not an incompatibility but is something you should be aware of should you notice it. In the event you get this, simply back off the S3200 filter a little - this should overcome it.

The next three parameters down the right hand side are the modulation inputs to the filter. The defaults for these are velocity: , Lfo2: and Env 2: (similar to the S1000 and S1100) respectively. These may be mixed and the range for each modulation input is the usual +/-50. You will note that for there to be any effect, the frequency: parameter should be set to something lower than 99.

With velocity: set to a high positive value, you may use velocity to control tone colour much like you would find on an acoustic instrument with louder notes yielding brighter sounds and, of course, vice versa. Lfo2: may be used for filter sweep effects such as flute tremolando or drastic resonant synth effects whilst Env2: is used for shaping the tonal dynamics of the sound and restoring lost harmonic movement due to looping. The multi-stages of Env2 allow some interesting possibilities as we shall see in a moment. The other options which you may select for modulating the filters are:

Modwheel: This works much like pressure and moving the modwheel will cause the filter cutoff to open and close.

Use this for phrasing brass parts, perhaps, or for special synth filter effects in a bass line or lead line.

Bend: This works like pressure and modwheel and allows you to open and close the filter by moving the pitch bend wheel or lever. This can be effective when bending up into a note as the filter will open and sound brighter.

Pressure: This may be used for expressive swells, particularly on brass sounds.

External: This can select from footpedal, volume and breath for control of the filter cutoff.

Key: Although selectable as a mod source, it is not that worthwhile because this is hardwired via the key follow parameter.

Lfo1: This allows you to emulate the natural tremolo of flutes, woodwind, brass and other such instruments when set to small modulation amounts. When set to large modulation amounts, classic synth filter sweeps can be achieved. That LFO 1 can also be modulated leads to some very interesting synth sounds and special effects.

Env1: On occasions, it is good to be able to have the filters tonal dynamics match those of the amplitudes. One easy way to achieve this is, instead of copying the amplitude envelope to the filters envelope, simply assign the amplitude envelope to the filter.

Env 3: This may be used as an alternative to Env 2.

! Modwheel: This and the other "! " controllers allow you to control the opening and closing of filter cutoff at the point of note on. They don't have any effect if these controllers change through the course of the note but only when the note is pressed.

THE SECOND FILTERS

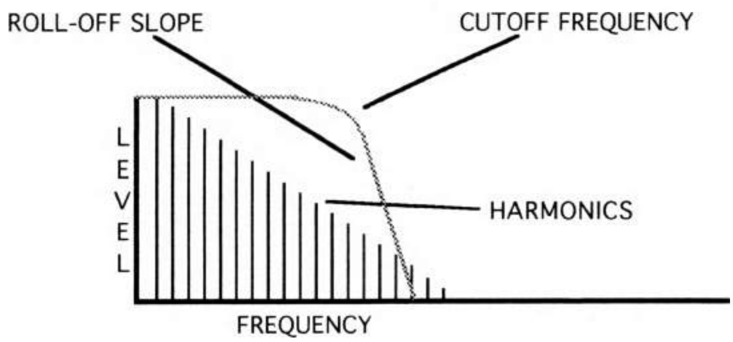
The second bank of filters is accessed by pressing FLT2. This will display the following screen:

```
FILTER2 KG:  1 ED:ONE   TEST PROGRAM   0%
C_0 - G_8       Filter2/Tone enable: ON
  frequency: 99    velocity > freq: +00
 key follow: +12       Lfo2 > freq: +00
  resonance:  0        Env2 > freq: +00
filter mode: LP       attenuator: +0dB
MAIN KGRP FLT1 FLT2 ENV1 ENV2 ENV3 TONE
```

Essentially, this looks very similar to the Filter 1 page except you will notice the extra filter `mode:` parameter that selects these filters' different modes.

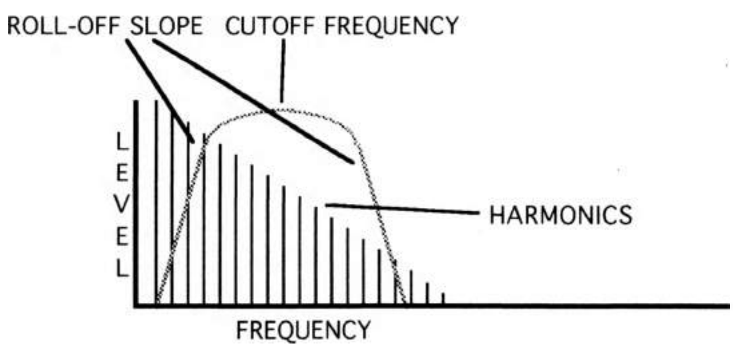
The second filter offers four different types of filters: lowpass, highpass and bandpass plus a special EQ mode.

The lowpass filter we have already seen and offers this type of response graph:



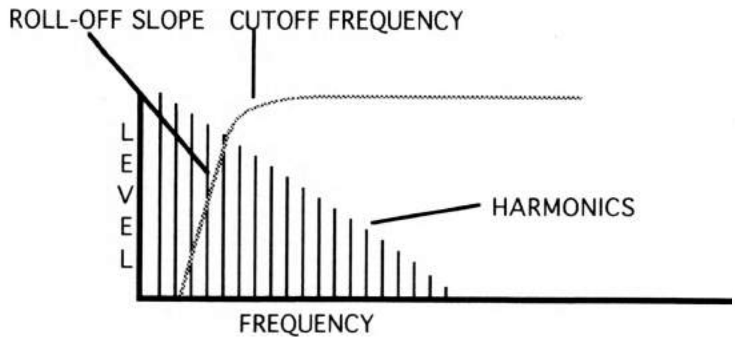
Here, high frequency components above the cutoff frequency are removed and only lower frequency components will pass through the filter. When the resonance control is increased, the area around the cutoff frequency is boosted to give synthesizer effects.

The band pass selection offer this type of response slope:



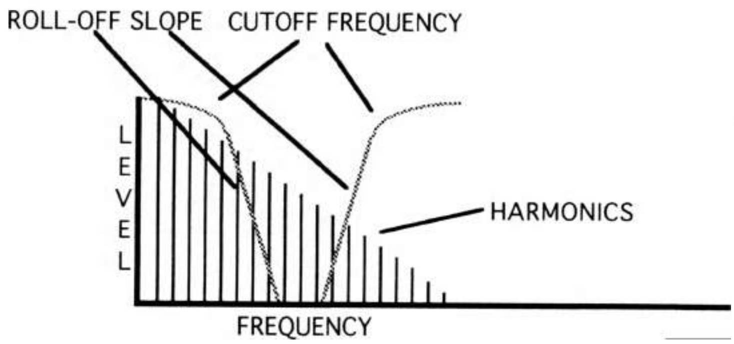
Here, frequencies below the cutoff and above are removed. You may simultaneously remove depth and top end using this selection. As the resonance amount is increased, so the width of the response slope gets narrower so that individual harmonics are emphasised.

The high pass filter offers this response slope:



In this example, you can see that low frequency components are removed whilst high frequencies pass through. This filter can be used to make sounds very thin and brittle. For example, this type of filter may be used effectively on an oboe sound or harpsichord sound. When the resonance is increased, the area around the cutoff frequency is boosted and so harmonics around that frequency will be emphasised.

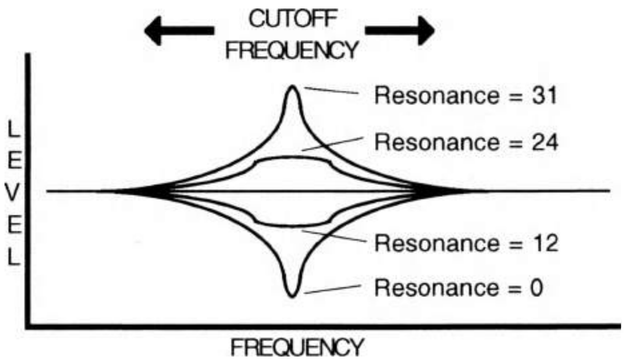
By using Filter 1 and Filter 2 together, by setting the cutoff frequency of Filter 1 lower than Filter 2, you can create what is known as a 'notch reject' filter whose response slope is like this:



Here, low frequency harmonics and high frequency harmonics are passed through but harmonics in the middle of the sound are removed. By setting the resonance control of both filters, you may create a sound with two resonant peaks. By modulating the two filters identically, the two cutoff points may track each other for the creation of very interesting sounds.

The final mode selection is quite unlike those shown above. Selecting EQ turns the second filter bank into a simple one band equaliser with variable frequency and resonant cut/boost that can be used for a variety of different effects. With the EQ selection, the 'straight' sound from Filter 1 is also passed through unaffected and you can use this EQ section to highlight specific frequencies in the sound. This filter is also able to be controlled by any of the modulation sources we have seen so far and using it with a high resonance setting in conjunction with any of these modulation sources, you may create interesting sounds not unlike phase shifting.

The response slope for the EQ selection is shown below:



With resonance at 16, the frequency response is flat but as the resonance is increased, the gain of the filter is boosted around that frequency. If the resonance is decreased, the gain is cut as the resonance gets sharper.

The fields on the FILTER 2 page are as follows:

The parameters across the top of the page follow the usual convention and allow you to select the keygroup for editing, select whether you wish to edit just one keygroup or all keygroups simultaneously and, of course, you can select another program for editing if you wish. The other fields on this page are:

`C_0 - G_8` This shows the current keygroups key range.

`frequency:` When lowpass, band pass and highpass filters are selected, this allows you to set the filters cutoff frequency. When EQ is selected, this control sets the EQ band's centre frequency..

`key follow:` Here you may set the keyboard to track the filter. This is so that you can achieve an even tone across the keyboard range. +12 is the default and this tracks the filter octave for octave - i.e. for every shift of pitch of one octave, there is an according shift in harmonics.

`resonance:` This allows you to sharpen the point at the cutoff frequency thereby emphasizing the harmonics at that point. The sound changes from a soft 'waaa' effect to the characteristic 'weeow' effect with high resonance settings. The range is 0-31. High resonance settings can be used for classic synth sounds and, simply by sampling raw synth waveforms (i.e. without using the synths filter, etc.), these may be used as the basis for some powerful synth sounds through these filters.

NOTE: *The resonance control functions differently when EQ is selected. When EQ is selected, a value of 16 is no cut or boost. Raising the resonance above 16 will boost the selected cutoff frequency and lowering it below 16 will cut the selected cutoff frequency. You will, therefore, experience a tonal change when you switch from LP, BP or HP to EQ. For example, if a value of 16 is set for the resonance on the other three filters, this will be flat when you select EQ. Similarly, if a value of 0 is set on the other three filters, this will cause the cutoff frequency to be cut when you select EQ.*

`filter mode:` This parameters allows you to select the filter mode of your choice and the selection possibilities are LP (lowpass), BP (bandpass), HP (highpass) and EQ. For details on these different filter types, please see the description given above.

`Filter2/Tone enable:` This allows you to bypass the second filter and tone section if you wish. You may use this parameter to quickly bypass the settings of the second filter and tone control section to make A/B comparisons.

NOTE 1: *With this parameter set to ON, the polyphony of the S3200 is restricted to 30 voices. If OFF is selected, the second filter and tone section are bypassed, and the S3200 may operate with 32 voices.*
NOTE 2: *The setting of this parameter relates to both the second filter and tone section.*

The next three parameters down the right hand side are the modulation inputs to the filter. The defaults for these are `velocity:`, `Lfo2:` and `Env3:` respectively. These may be mixed and the range for each modulation input is the usual +/-50. You will note that for there to be any effect, the `frequency:` parameter should be set to something lower than 99.

With `velocity:` set to a high positive value, you may use velocity to control tone colour much like you would find on an acoustic instrument with louder notes yielding brighter sounds and, of course, vice versa. `Lfo2:` may be used for filter sweep effects such as flute tremolando or drastic resonant synth effects whilst `Env3:` is used for shaping the tonal dynamics of the sound and restoring lost harmonic movement due to looping. The multi-stages of `Env3:` allow some interesting possibilities as we shall see in a moment. The other options which you may select for modulating the filters are:

`Modwheel:` This works much like pressure and moving the modwheel will cause the filter cutoff to open and close. Use this for phrasing brass parts, perhaps, or for special synth filter effects in a bass line or lead line.

`Bend:` This works like pressure and modwheel and allows you to open and close the filter by moving the pitch bend wheel or lever. This can be effective when bending up into a note as the filter will open and sound brighter.

`Pressure:` This may be used for expressive swells, particularly on brass sounds.

`External:` This can select from footpedal, volume and breath for control of the filter cutoff.

`Key:` Although selectable as a mod source, it is not that worthwhile because this is hardwired via the `key follow:` parameter.

`Lfo1:` This allows you to emulate the natural tremolo of flutes, woodwind, brass and other such instruments when set to small modulation amounts. When set to large modulation amounts, classic synth filter sweeps can be achieved. That LFO1 can also be modulated leads

to some very interesting synth sounds and special effects.

Env1: On occasions, it is good to be able to have the filters tonal dynamics match those of the amplitudes. One easy way to achieve this is, instead of copying the amplitude envelope to the filters envelope, simply assign the amplitude envelope to the filter.

Env 2: This may be used as an alternative to Env3 perhaps.

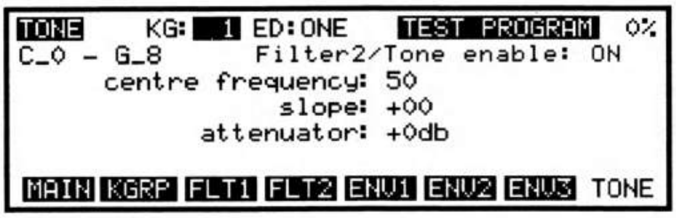
! Modwheel: This and the other "! " controllers allow you to control the opening and closing of filter cutoff at the point of note on. They don't have any effect if these controllers change through the course of the note but only when the note is pressed.

The final parameter on this page is the attenuator: field and allows you to switch in a -6dB pad. This is included to overcome the possibility of distortion should the resonance be set quite high, boosting the sound into overload.

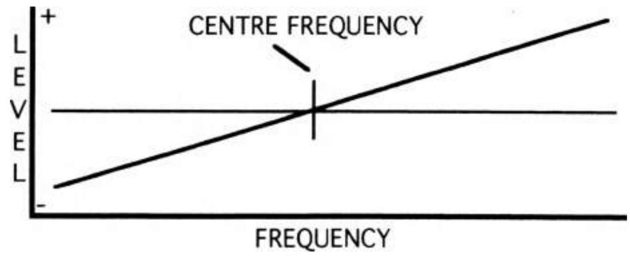
NOTE: *The attenuator should be enough to overcome distortion that may result from very high resonance settings. If, however, it is not, then you will have to turn the program level down in the OUT page.*

THE TONE PAGE

The S3200 also features a very simple but effective tone control which is accessed via F8 - TONE. Pressing this will display this screen:



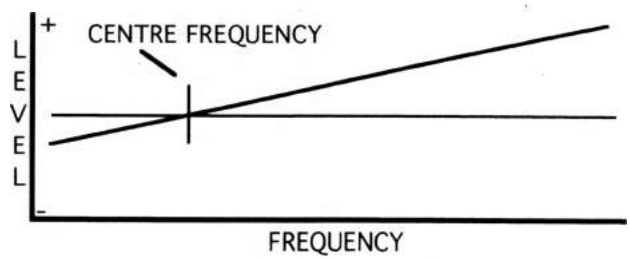
The TONE section can best be described as 'spectral tilt'. If you imagine a see-saw, the bench is the `slope:` parameter and the fulcrum over which it rocks is the `centre frequency:` parameter. For example, when the controls are set as shown in the above screen diagram, the response would be flat but with a setting of 50 for the centre frequency parameter and +50 for the slope parameter, you would have a response graph something like this:



CENTRE FREQUENCY=50 SLOPE=+50

Here, bass frequencies are attenuated whilst high frequencies are boosted. Setting the slope parameter to -50 would reverse the angle, cutting high frequencies and boosting LF.

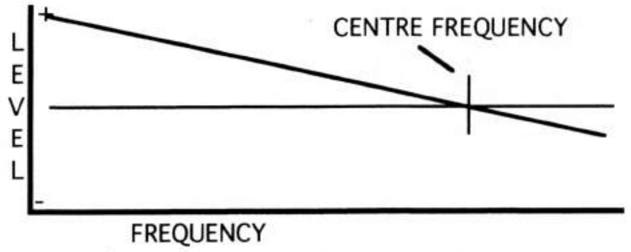
By setting the centre frequency lower, you may create a tone response something like this:



CENTRE FREQUENCY=25 SLOPE=+50

Here, some bass frequencies are cut but some mid and a lot of high frequencies are boosted.

Setting a higher centre frequency value with a negative slope value may produce something like this:



CENTRE FREQUENCY=75 SLOPE=-50

Here, bass frequencies and some mid range components are boosted whilst high frequencies are cut.

The main purpose of this section is to be able to gently remove unwanted noise from a sound. For example, you could use it to remove some mains hum from a sound or some hiss or other high frequency noise. Used together, the second filter and the tone section can be used very effectively to 'clean up' samples.

You may also use the TONE section to balance the tone of an instrument across the keyboard range - for example, you may give a set of string samples more bottom end to emphasize the bass instruments.

Of course, the TONE section can also be used as a simple EQ and, because it is available for each keygroup, you could use it to add depth to a kick drum or snare drum or to add some top end to cymbals, hi-hats and snares. Other sounds may benefit from some simple tonal modification in this section too. Experiment!!

The `attenuator:` parameter allows you to switch in a -6dB pad. This is included to overcome the possibility of distortion should the resonance of the second filters or the slope of the TONE section be set quite high, boosting the sound into overload.

The `Filter2/Tone enable:` parameter allows you to bypass the tone control section.

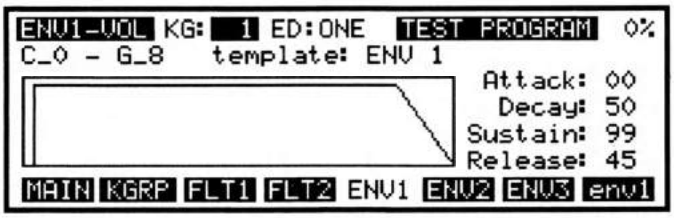
NOTE: *The attenuator and Filter2/Tone enable fields are a duplication of the ones found in the FILTER 2 page. If you switch them on or off in this page, you will notice that they will be set the same in the FILTER 2 page. The converse is also true - switching them on of off in the FILTER 2 page will have the same effect in the TONE page.*

THE ENVELOPE GENERATORS

The S3200 has three envelope generators. Two of them are multi-stage offering control of four rates and levels whilst the other is a simple ADSR type hardwired for amplitude control. To set the envelopes for the sound you have two ways in - you can either go to the envelope pages via the KGRP page or, if you are in the filter page, you have direct access to them there for convenience. Either way, let's have a look at ENV 1.

ENV1 - SHAPING AMPLITUDE

However you arrive at this page, whether its through the KGRP page or directly from the filter page, the screen looks like this:



Here we have the normal parameters across the top of the page where you may select your keygroup, whether one or all keygroups are being edited and the program name. Also, beneath that you can see a graphic representation of the envelope. The keyspan is also shown. The other parameters are:

template: This calls up a series of preset envelope templates that have been set within the software of the S3200. You can use these to get close to the type of envelope you are after and then maybe fine tune them afterwards if needs be. ENV1 is the 'manual' envelope - i.e. the one you can program yourself. If you edit a preset envelope, you will note that it immediately become ENV1, the programmable envelope.

You will notice that any envelope you create is not lost when you select a preset - ENV1 (your own envelope) is always retained as you scroll through the list of available envelopes although ENV1 will be lost if you edit a preset.

***NOTE**: Should you select a preset and then leave this page, when you return, you will note that the template is renamed and becomes ENV1.*

Attack: This sets the time the envelope will take to reach full level.

Decay: This sets the time it will take to reach the sustain level.

Sustain: This sets the level at which the note will sustain while a key is held.

Release: This sets the time it takes for the sound to fade away after the note has been released.

This forms the basis of an ADSR envelope generator for shaping amplitude. This envelope generator is hardwired to amplitude control and so always sets the sounds overall envelope.

These are the most commonly used parameters in the envelope. Other, less frequently used controls, are on a second page should you need them.

Pressing env1 calls up this screen:

```
ENV1-VOL KG:  1 ED:ONE  TEST PROGRAM   0%
C_0 - G_8        velocity>attack: +00
                velocity>release: +00
            off velocity>release: +00
             key>decay & release: +00
                     attack HOLD: OFF
MAIN KGRP FLT1 FLT2 ENV1 ENV2 ENV3 env1
```

Once the basic envelope has been set up, these other factors can be used to affect the speed of the envelope.

velocity > attack:

This is variable from +50 to -50, and determines the amount by which the attack speed will be changed depending on the Note On velocity. A positive value will increase the attack time if the key is pressed fast, while a negative value will slow down the attack rate if the key is pressed fast. Setting a positive value here is the most commonly used way of using this parameter and is useful for imitating the characteristics of some acoustic instruments (for instance, most wind instruments have a faster attack rate when played loudly).

velocity > release:

This is used to vary the release rate relative to the Note On velocity value (+50 to -50). Again, a positive value will increase the release rate relative to the Note On velocity, and a negative value will shorten the release time.

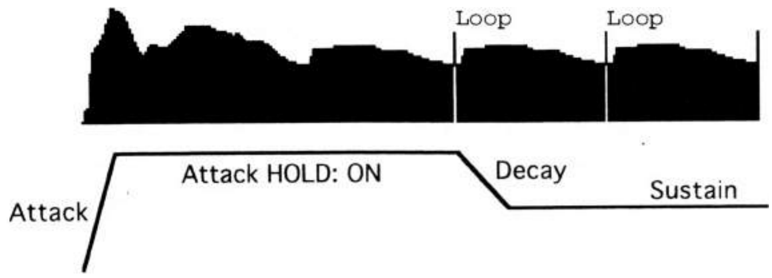
off velocity > release:

This is possibly more relevant to natural-sounding playing. The MIDI specification allows for Note Off velocity as well as Note On velocity. Though some keyboards do not accept or transmit this, assuming a mean value of 64, all AKAI keyboards provide a full implementation of this function. The speed with which the key is released can be used here to affect the release rate (positive values mean that a fast release lengthens the release rate, and vice versa).

This allows more expressiveness and realism, but demands a slight re-learning of keyboard technique (similar to an acoustic piano).

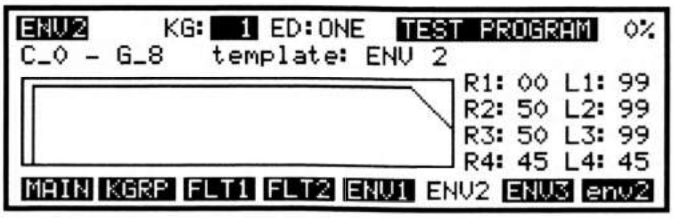
`key > decay & release:` This allows you to control the amount by which the key position affects the decay and release rates. Setting this to a negative value means that the higher the note played on the keyboard, the shorter the decay and release times (similar to most acoustic instruments). This can be used to good effect on marimbas and other such percussive sounds and can be effective on piano sounds too. Setting this parameter to a positive value will reverse this effect.

`attack hold:` can be set to ON or OFF. When ON, the attack portion of the envelope will be held until looping begins, and when OFF, the envelope will continue along the set values, regardless of loop settings. I.e:



ENV2 - SHAPING THE FILTER

Access to ENV2 is also via the KGRP or FILT page. However you arrive there, the screen looks like this:



This is a 4-stage envelope generator with 4 rates going to 4 levels. Basically, Rate 1 goes to Level 1, Rate 2 to Level 2, Rate 3 to Level 3 (which is also the sustain) and Rate 4 goes to Level 4. Again you have a choice of templates from which to choose a variety of preset envelopes (probably a bit more necessary in light of the added complication of a multi-stage envelope) and this works on the same principle as ENV1's templates except that there are more of them.

Some of the possible envelope shapes you can create using envelope 2 are shown below:

The second page of envelope parameters can be accessed by pressing env2. You will receive this screen:

```
ENV2      KG:  1 ED:ONE  TEST PROGRAM   0%
C_0 - G_8       velocity>R1: +00
                velocity>R4: +00
            off velocity>R4: +00
                 key>R2 & R4: +00
          velocity>envelope: +00
MAIN KGRP FLT1 FLT2 ENV1 ENV2 ENV3 env2
```

Here we have similar parameters to ENV1.

`velocity > R1:` This sets how much velocity will determine the speed of rate 1.

`velocity > R4:` This sets how much the note-on velocity will affect the speed of rate 4.

`off velocity > R4` This sets the amount by which MIDI note off velocity will affect the speed of rate 4.

`key > R2 & R4:` This will set how much key position will affect both rate 2 and rate 4.

`velocity > envelope:` This sets how much velocity will control the modulation output of ENV2. This parameter can be used very effectively in regulating dynamics through key velocity. All parameters' ranges are +/-50.

ENV3

The S3200 has a third envelope generator which is exactly the same as ENV2. This has no defined function (although its default assignment is to control the cutoff frequency of FILTER 2) but may be freely assigned to anything you wish. Typically it may be used to control FILTER 2 separately but may also be used to control pitch, panning, LFO1 rate, etc., especially if ENV2 is busy doing other things. Pressing ENV3 will display this screen:

```
ENV3      KG:  1 ED:ONE  TEST PROGRAM   0%
C_0 - G_8   template: ENV 2
                                  R1: 00 L1: 99
                                  R2: 50 L2: 99
                                  R3: 50 L3: 99
                                  R4: 45 L4: 45
MAIN KGRP FLT1 FLT2 ENV1 ENV2 ENV3 env3
```

Because its parameters are identical to those in ENV2, please refer to the explanation ENV2. A second page of envelope parameters is also available and is accessed by pressing env3. This will display this screen:

```
ENV3      KG:  1 ED:ONE  TEST PROGRAM   0%
C_0 - G_8       velocity>R1: +00
                velocity>R4: +00
            off velocity>R4: +00
                 key>R2 & R4: +00
          velocity>envelope: +00
MAIN KGRP FLT1 FLT2 ENV1 ENV2 ENV3 env3
```

Again, because the parameters are identical to ENV2, please refer to that for more information.

USING THE FILTERS AND ENVELOPE GENERATORS

One of the inherent problems of sampling is that, because of memory limitations, it is usually necessary to loop a sample. This often has the effect of reducing (or even removing!) the sound's natural dynamics making the sample more like a snapshot than a movie.

To overcome this, however, we can use the filters and the envelope generators to restore some of those attributes.

By bringing the filter 1 cutoff frequency down and using note-on velocity as a controller, you can emulate the characteristics of most, if not all acoustic sounds where loud notes are brighter in tone than quiet ones. Controlling the output of ENV2 or ENV3 using velocity and applying that to the filter is another way of doing this. Furthermore, another natural property of sound is that during a note, the tonal quality changes. We can use envelopes and low frequency oscillators to restore some of those qualities. Usually, the filter will use the multi-stage ENV2 or ENV3 as its controller because tonal changes are usually more complex than amplitude changes in acoustic sounds. This envelope can also be used to imitate such things as brass growls - to actually sample a brass growl would not only take up memory but would speed up and slow down as you play it across the keyboard. ENV2 or ENV3 could be applied to an ordinary brass sample and set so that Rate 3 swells back up from a low Level 2 to not only recreate this effect but also to keep the swell at a constant rate across the keyboard. Pressure may also be used for the same effect although you would be controlling the growl - this may be preferable to the 'automated' quality a preset envelope rate would have.

The second bank of filters allow even more extreme tonal processing to take place. These filters may be used in conjunction with the filter 1 simply to 'clean' up the sound, removing unwanted artefacts from the sound. They may also be used very creatively.

For example, the highpass filter may be used to more accurately replicate the thin tone of an oboe or harpsichord as could the bandpass filter. Using two lowpass filters in series allows you to create a notch filter to remove mid frequencies. If resonance is applied to both lowpass filters, you may set two resonant peaks. This can be useful in re-creating certain sounds that actually have more than one resonant peak. You could try assigning, say, Env 2 to modulate both. By setting one filter to have a positive going modulation and the other to have negative going modulation, you can create some interesting tonal variation during the course of a note - in fact, careful control in this way can yield some very haunting vocal articulation effects.

Of course, you can combine low and high or band pass filters and, when you apply modulation, some truly bizarre effects are possible. With so many permutations, it is impossible to be give specific guidelines. The trick is experimentation!

Of course, once you are in the realm of sampling synth waveforms and processing them through the filters, you are in different territory but anyone who has used an analogue synth will feel instantly at home with the S3200's filter and envelope section. The only difference here, however, is that instead of relying on a handful of waveforms, any sampled sound may be used as the source. It is here that the modulation possibilities can be used to good effect in the creation of new sounds (and the recreation of a few classic old synth sounds too!)

Having the filters, envelope generators and comprehensive modulation facilities in the S3200 means that instead of having to sample an entire synth sound, you can simply sample the 'raw' waveforms and apply all the other synth processing in the S3200. There are several ways this can be done - you could either sample multiple detuned oscillators or you could sample individual oscillators and then layer them in the S3200. This might be preferable in a way as single waveforms can easily be looped and take up virtually no memory space (*). With digital synths, you can take the basic waveform material and build up a huge array of waveforms to use as the basis of your synth sounds.

** **NOTE**: Sampled waveforms do not detune in quite the same way as analogue synths. On analogue synths, there are all kinds of pleasant distortion artefacts that give the sound character - in some cases it is best to sample that distortion. Note also that sampled waveforms transposed up and down the whole range of the keyboard do not sound quite the same as 'the real thing' so it is probably best to multi-sample these for best results.*

KEYGROUP PITCH/AMPLITUDE MODULATION

The final page in PROGRAM EDIT is where you may assign modulation to pitch and amplitude for individual keygroups. This is accessed via the KGRP page by pressing PTCH. You will receive this screen display:

```
PITCH/AMP KG:  1 ED:ONE TEST PROGRAM  0%
C_0 - G_1
                 LFO1 > pitch: +50
                 Env2 > pitch: +00

         Velocity > loudness: +00
MAIN KGRP SPAN FILT ENV SMPL PTCH
```

Along the top of the screen we have the usual parameters for selecting the keygroup and the program. The other parameters on this page are:

`LFO1 > pitch:` This is a fixed, preset assignment that routes the LFO to pitch. This is done to maintain compatibility between the S1000/S1100 and the S3200. It is also done to maintain ease of use when setting up vibrato.

The range for this parameter is +/-50 allowing inverted pitch effects to be created (especially useful when using square and sawtooth waves) and the default for this parameter is +50. This means that the modwheel is always active for vibrato without you needing to program or set anything up on LFO1. It also means you only have to set a value in the depth field of the LFO1 page to have a constant vibrato.

***NOTE:** If you wish to use LFO1 for some other modulation application such as filter sweeps, panning, etc., you will need to turn this value to 00 otherwise pitch will also be modulated unless, of course, that's what you want.*

It is not possible to route any other controllers in this field.

`Env2 > pitch:` This is a freely assignable modulation input and any source may be selected here. Env2 is selected as the

default again as means of ensuring compatibility between the older samplers and the S3200. Feel free to route anything you like to this field. Some suggestions are:

Pitchbend: Although there is a global pitchbend function in the modulation pages, you might like to use this to individually bend keygroups separately. The values for pitchbend range are:

+04	1 semitone	+09	1 tone
+13	minor 3rd	+17	major 3rd
+21	fourth	+26	6 semitones
+30	fifth	+34	8 semitones
+38	9 semitones	+42	10 semitones
+46	11 semitones	+50	1 octave

Be sure to turn the pitchbend parameters to 0 in the modulation pages unless you wish to add the above values to the pitchbend set there.

Modwheel: Use this instead of pitchbend. The values for pitchbend are the same as above. Be sure to turn modulation to +00 in the LFO1 > Pitch: field above unless you wish the bent note to also have vibrato.

Pressure: Use this instead of either of the above. The same values apply

External: Use this instead of the above when using the footpedal or breath control possibilities for pitchbend.

Velocity: Use this so that differing velocities will affect pitch. This may be useful on some percussive sounds which have a different pitch for each note - for example, an African talking drum or pedal tympani.

key: Although this may appear pointless as the keyboard is also routed, by setting a negative value here you may set up microtonal scales. By setting a positive value you may extend the usual keyboard tuning.

Lfo2: Add this to LFO1 for a more varied vibrato with ensemble sounds. Alternatively, setting LFO and LFO2 accordingly, you may create some odd special effects. LFO2 could also be set to provide a square wave octave jump whilst LFO1 provides vibrato as normal. Many possibilities exist.

Env1: Use this to create pitch sweeps that vary according to the sounds overall level.

Env 3: This may be used as an alternative to Env2, especially if Env2 is tied up with filter sweeps.

! Modwheel: Use this and the other "! " controllers to affect pitch at the point of note-on.

`Velocity>loudness:` This is another freely assignable modulation field that affects individual keygroups' loudness and `velocity > loudness:` has been 'inherited' from the S1000 and S1100 to ensure compatibility of sound disks. You may of course, assign anything you want here. Some ideas are LFO1 or LFO2 for tremolo effects (try layering keygroups and setting one positive and the other negative to create undulating crossfades between two samples). Any of the MIDI controllers such as modwheel, pressure, bend, etc., may also be used to control loudness.

***NOTE**: The LOUDNESS modulation here is different from that we saw earlier in the OUT page. The OUT page is master control for the level of the whole program - this parameter in this page is applicable to individual keygroups.*

CONCLUSION

As you can see, setting up programs is not radically different from setting up sounds on a synthesizer except that the assignable modulation parameters make it much more versatile. The same principles of envelope shape, LFO, etc., apply to both. The main difference is that you start with your own samples rather than preset waveforms thus giving you an almost infinite flexibility in the sounds you can create.

If you are already familiar with the S3200's predecessors, we hope you like the refinements that have gone into the S3200's EDIT PROGRAM functions, refinements we have been able to include thanks to the on-going feedback we have received from our customers.

MIDI

When you first press the MIDI mode key, the BASIC MIDI CHANNEL CONTROL page is entered. This shows a number of parameters which affect the MIDI response of the whole instrument.

```
BASIC MIDI CHANNEL CONTROL
            program select: OMNI
                global OMNI:  ON
                          :
     external controller: BREATH

 CHAN FILT PPMs RCVE TRAN EXCL SCSI
```

The parameters are as follows:

program select: This allows you to enable or disable Program Change commands. The choices are OFF, 1 to 16 and OMNI. When OFF is selected, then program change commands will be ignored. Selecting a number between 1 and 16 switches program change enable on and sets the MIDI channel you wish to use for program change. Selecting OMNI will cause program change commands received on any MIDI channel to select programs. This parameter defaults to OMNI so MIDI program change commands will be accepted on all MIDI channels.

global OMNI: This allows you to switch OMNI reception of MIDI channels (i.e. reception of MIDI on ALL channels) on or off. This affects the whole unit and also overrides the setting in program select enable: . For convenience, you may prefer to have this switch OMNI to ON so that you don't have to worry about which MIDI channel to use but, especially when sequencing multi-timbrally, you will need to turn OMNI to OFF.

external controller: This allows you to select the external MIDI source used in the PROGRAM ASSIGNABLE MODULATION system used in programs. Whatever you select here becomes one of the choices you can make when assigning modulation sources in a program.

Normally, there is little to do in this screen as all the defaults have been sensibly chosen and are shown above. You will note that these parameters are saved to disk whenever you perform a VOLUME save. They are not saved when performing any other type of save.

MIDI FILTER

Pressing the FILT key (F2) will display this screen:

```
MIDI RECEIVE FILTERS   +on  -off
CHAN: 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 all
  ON: + + + + + + + + + +  +  +  +  +  +  +  <
WHLS: + + + + + + + + + +  +  +  +  +  +  +  <
PRES: + + + + + + + + + +  +  +  +  +  +  +  <
LOUD: + + + + + + + + + +  +  +  +  +  +  +  <
CHAN FILT PPMs RCVE TRAN EXCL SCSI
```

This page allows you to filter out specific MIDI information. When you enter this page, the cursor will be at the top left of a grid of '+' signs, in a long rectangular box. You can use the cursor keys to move to any point on the screen.

In this grid, the columns represent MIDI channels (1-16), and the rows represent MIDI information which the S3200 will accept or filter. The first row, 'ON:', affects the S3200's receive capabilities for all information on that channel, the next, 'WHLS:', refers to the pitch and modulation wheels, the third line, 'PRES:', refers to aftertouch, and the last line, 'LOUD:', refers to an external MIDI volume control (controller 7). The last column in each row, 'all', will affect the appropriate information for all MIDI channels. The '+' signs mean that the S3200 accepts this information and '–' means that this information is filtered out.

If you turn the DATA control counter-clockwise, the '+' which the cursor covers will change to a '–'. If you make this change in the ON row, all '+' signs in the same column below will change to a '–' and you will see a column of '–'. If you make this change in the 'all' column on the right, the parameter for all MIDI channels will be changed and you will see a row of '–'. The top right corner of the display (ON/all) is a special case - all parameters will be changed which may be useful for resetting the whole screen.

By using this filter, you can control the response of the S3200 to MIDI events. By filtering out aftertouch on a percussion program where it is not needed, for instance, you can improve the response of the S3200 when a lot of MIDI data is received.

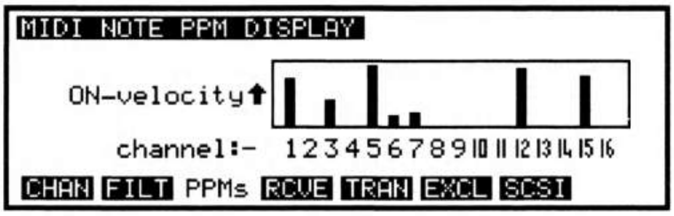
PPM PAGE - MONITORING MIDI

Pressing the PPMs key (F3) will display this screen:

```
MIDI NOTE PPM DISPLAY

    ON-velocity↑ |▌                  |
       channel:-  12345678910111213141516
CHAN FILT PPMs RCVE TRAN EXCL SCSI
```

This page has no parameters. Instead, it provides a real-time display of all Note On information received on the 16 MIDI channels. The higher the bar on the display, the greater the velocity of the received note. This page is called 'PPMs' because it simulates the behaviour of audio bar-graph Peak Program Meters.

Under normal circumstances when playing the S3200 from a MIDI keyboard, you will receive a display such as is shown above with the bar graph showing incoming MIDI on the selected channel but, when sequencing multi-timbrally on several channels, you will see a display such as:



This is a very useful page that allows you to track down any problems you may be experiencing when sequencing. For example, if a part isn't sounding, you can check if the S3200 is receiving MIDI on its channel. If it is, then it may be some other problem such as wrong output assignment, channel fader on the mixer not open, the sound hasn't loaded, etc..

MIDI RECEIVE PAGE - ANALYSING MIDI

Another MIDI receive monitor is available in the RCVE page. Here, you can monitor other types of MIDI information received by the S3200.

```
MIDI RECEIVE MONITOR        CHAN: OM
           ↑                         dot
           ↑                        means
           ↑                       running
           ↑                       status
latest >
CHAN FILT PPMs RCVE TRAN EXCL SCSI
```

Again, this is especially useful if you are faultfinding on your MIDI system and you're convinced that you're transmitting note information on a certain channel, but the S3200 isn't responding. Information from the MIDI IN is displayed, and a channel filter may be set at the top of the screen (OMNI or 1 to 16).

When idle (i.e. not receiving MIDI), the screen shown above is displayed - when receiving MIDI, something like the following is shown:

```
MIDI RECEIVE MONITOR        CHAN: OM
         ↑ F_4  ON   VEL= 80  1      dot
         ↑ A_4  ON   VEL= 65  1     means
         ↑ C_5  ON   VEL=110  1    running
         ↑ C_5  OFF  VEL= 47  1    status
latest   > F_4  OFF  VEL= 45  1
CHAN FILT PPMs RCVE TRAN EXCL SCSI
```

Here you will see a constantly changing display as notes are received. If any performance controls are used such as mod wheel or pressure, these too will be shown.

If the information is not displayed on this monitor, the information is not reaching the S3200. Check your connections or the output channel of the transmitting equipment. If the information appears to be correct, but no sound or unexpected sounds are being produced, then the fault may lie in the MIDI setting of the program(s). You may discover that the piano track of the sequencer is playing the drum samples, for example.

You may select to view particular channels using the CHAN: field. This defaults to OM (short for OMNI so you can view all channels but you may select individual channels 1-16 if you only wish to monitor a specific channel.

TRANSMIT TEST PAGE - SETTING THE ENT/PLAY KEY

Pressing the TRAN soft key (F5) will display this screen.

```
MIDI NOTE TRANSMIT TEST

          channel:  1
             note: C_3
         velocity: 127
                                  ┌SEND┐
CHAN FILT PPMs RCVE TRAN EXCL  ON   OFF
```

In the TRANS page, you can produce a test transmission of a MIDI note, and set the channel, key and velocity of the note to be transmitted, using the ON and OFF keys (F7 and F8). This key value and velocity will also be used by the key when testing samples and programs using the ENT/PLAY key.

***NOTE**: When in EDIT SAMPLE mode, the raw sample is always played at its base pitch (i.e. the pitch was sampled at) and not at the note value set here.*

MIDI DATA DUMPS

Samples can be transferred between the S3200 and other samplers via MIDI and this is done in the EXCL page:

```
MIDI EXCLUSIVE channel:  1 (trans & rec)
  type of transmission: ALL PROGRAMS
       sample protocol: STANDARD
        single program: STRINGS 1
         single sample: STRING C4
sample number override:  2
CHAN FILT PPMs RCVE TRAN EXCL SCSI SEND
```

To perform a MIDI data dump, you must make a MIDI loop (MIDI OUT of the S3200 connected to MIDI IN of the other sampler, and vice versa). This is necessary because of the way in which MIDI sample dumps are performed (in computer terms, a handshake protocol with error detection/correction).

Though the S3200 is a 16-bit sampler, it can accept samples from other samplers, including those from other manufacturers which use a lower bit resolution. If transmitting to another sampler which uses fewer bits, the S3200 simply truncates the low bits during the transfer. Instead of another sampler, samples can be transmitted to and accepted from other devices (such as computers) which are capable of storing and/or editing sample data with the appropriate sample editing software. However, in this section we will always refer to the other device as a 'sampler'. The fields are as follows:

channel: — This does not refer to a MIDI channel, but a 'logical channel' used in System Exclusive protocol. Both samplers must be set to the same channel for transfer to take place.

type of transmission: — This parameter refers to what will actually be transmitted over MIDI. This can be ALL

PROGRAMS, ALL SAMPLES, SINGLE PROGRAM, SINGLE SAMPLE, and DRUM SETTINGS. The meanings of these values should be self-explanatory.

sample protocol: Two protocols for sample transfer are available. STANDARD conforms to the MIDI sample dump standard and will dump only samples across and S3000, which is a superset of the MIDI sample dump standard which will dump everything including loop and other data. Use the S3000 setting only if you are transferring data between two S3200-compatible machines.

single program: If you have selected SINGLE PROGRAM transmission, this parameter allows you to specify the program which will be transmitted.

single sample: If you have selected SINGLE SAMPLE transmission, this parameter allows you to specify the sample which will be transmitted.

sample number override: You can override the default sample number (based on the order in which samples appear in the S3200's memory) with this parameter.

PERFORMING A MIDI DATA DUMP

When all the parameters are set up, press the SEND key to initiate transmission. Once the handshake protocol has been successfully initiated between the two devices, data transfer will take place. A new soft key, ABORT, will appear. Press this if you want to terminate the transmission prematurely.

There is no receive key on the S3200 as reception of bulk data will automatically take place once a remote device initiates the dump protocol.

***NOTE**: It is quite likely (if not certain) that sample editors will not work if you use the S3200 protocol because the editor will not have the ability to recognise the new file header information present because of the new features in the S3200. No doubt, manufacturers of these editors will soon upgrade their software to overcome this. In the meantime, you should use the standard MIDI sample dump protocol to exchange sounds between your editor and the S3200.*

MIDI VIA SCSI

The final soft key in this mode, F7, calls up the SCSI screen:

```
SCSI COMMUNICATION
          MIDI via SCSI: OFF
          local SCSI ID: 6
         remote SCSI ID: 6

CHAN FILT PPMs RCVE TRAN EXCL SCSI Sres
```

On this page you can enable or disable MIDI bulk data transmission along a SCSI buss, rather than the MIDI connections, and set the SCSI IDs for both the S3200 and the other SCSI device. The other SCSI device can be another of the S3200 series or a personal computer (equipped with the appropriate software). Values for SCSI device IDs can be from 0 to 7 and the two devices must have different SCSI numbers, otherwise there will be a conflict on the SCSI buss as two devices try to share the same ID. SCSI transmission of MIDI is much faster than normal MIDI data dumps which can take an awful long time!!

The soft key Sres allows you to reset the SCSI of your S3200 should some problem occur. When the SCSI buss gets busy as is the case when using a hard disk or other such SCSI device and especially when several SCSI devices are sharing the same buss, you can occasionally get SCSI errors. This is not a fault of the S3200 but will happen on any busy SCSI system. If you get this problem, use this key to reset the buss.

DISK MODE

This mode allows you to perform a number of disk-related operations, over and above those which you can perform from the SELECT PROG mode. Pressing the DISK mode key will display a screen something like the following:

```
LOAD FROM DISK: FLOPPYH vol: NOT NAMED
free memory: 100%     |STRINGS 1       P  0%
 free P/K/S 1012      |STRINGS 2       P  0%
type of load:-        |SLOW STRINGS    P  0%
 ENTIRE VOLUME        |STRING C2       S  5%
progs:  3 samps:  7   |STRING C3       S  6%
 LOAD SAVE REN  DEL  HDSK FORM  CLR  GO
```

LOADING FROM DISK

The S3200 allows you to load samples, programs, programs together with their associated samples, drum settings, Qlists, FX and Song files and operating systems from disk. This flexibility helps you pick and choose the sounds you need for a session or performance with the minimum of trouble.

After inserting a disk with data on it, press the LOAD soft key. If you haven't inserted a disk when you go to the disk page or press load, you will receive this message:

```
NO DISK !
```

If the disk is unformatted (or has become seriously damaged in some way) the S3200 will tell you:

```
UNREADABLE FORMAT ! or unformatted?
```

In this case, you will need to format the disk (see below).

***NOTE**: If this disk is able to be read by another S3200, S3000 or S2800, then this might indicate that there is a problem with your sampler's disk drive. Please contact your nearest Akai authorised service department.*

A floppy disk can only contain one volume, and may be formatted as high or DD. When you insert a new disk into the drive, if that disk's density (high or low - HD or DD) is different from the density of the last disk inserted, the S3200 will try an alternative density. The density of the disk currently inserted will be displayed as FLOPPYH or FLOPPYL at the top of the screen.

***IMPORTANT NOTE**: On the S1000 and S1100, it is possible to format DD disks to a high density format. This is not possible on the S3200. Furthermore, the S3200 cannot read DD disks that have been formatted on an S1000/1100 to a high density format. You will need to first load these sounds into an S1000/1100, resave them onto high density disks (formatted to high density, of course) before they can be used in the S3200. High density disks have a hole on the right hand side which is used by the disk drive to detect that it is a high density disk. If the S3200 does not 'see' this hole, it assumes it is a DD disk and so searches for a DD format. If it doesn't find it (i.e. because the disk is high density format), it cannot read it.*

If you have a hard disk fitted, you can select 'HARD-: ' at the top of the screen. You will receive a screen display such as:

```
LOAD FROM DISK: HARD-:A vol: NOT NAMED
free memory: 100%   |STRINGS 1       P  0%
 free P/K/S 1012    |STRINGS 2       P  0%
type of load:-      |SLOW STRINGS    P  0%
 ENTIRE VOLUME      |STRING C2       S  5%
progs:  3 samps:  7 |STRING C3       S  6%
 LOAD SAVE REN DEL HDSK FORM CLR  GO
```

This shows we have selected the hard disk and have selected partition A.

Hard disks can be divided into partitions (see the section on formatting for full details of how partitions are arranged) and the partition letter can be selected following the selection of HARD-: . Note that there will be a slight delay after choosing a partition while the partition is selected and read by the S3200. One partition on a hard disk can contain up to 128 volumes and each volume can contain up to 512 'items' - that is, combinations of programs, samples, effects files, Qlists, etc., and you can select the volume from which you want to load data in the next parameter field - vol: .

A list of all files (programs, samples and drum settings) will be displayed on the right side of the page. Programs have a 'P' beside their name, samples have an 'S', and drum input settings have a 'D'. Effects files have an 'X' beside them and Qlists have a 'Q'. Song files created in the disk record functions (see later) have a 'T' alongside them (T stands for TL or 'take list'). Floppy disks or hard disk volumes that contain S1000 or S1100 samples will have a '1' after these letters to signify they are from this series of sampler. They may be freely loaded into the S3200 without any problems.

Beside the type of file on the disk, there is also a percentage number along side it, which gives the amount of space that this file will take when loaded into memory. Programs, Qlists and effects files will usually show 0%.

The left hand side of the display shows you how many programs, keygroups and samples are free in memory (you may have a total of 1,022 'items') and at the bottom it displays how many programs and samples are currently in memory.

If you cannot see the file you want to load, move the cursor to the list of files and scroll up and down to display all the files on the disk. If the file you want is not on the disk, insert another disk and press LOAD to re-read the disk. If you have a hard disk fitted, then you can choose another volume to read. When you know that you have the right disk or volume, you can proceed.

The parameter under the type of load:- message can take a number of values. These are described below. When you have selected the appropriate value, you can press CLR or GO. CLR (CLEAR) will delete all programs and samples from memory, and then load the chosen file(s) from disk. You will receive this prompt to check you want to clear the memory:

```
                    |STRING C3       S  6%
CLEAR MEM THEN LOAD ?? confirm  NO   YES
```

and you should make the appropriate response.

Pressing GO will try to load the chosen file(s) into memory without deleting anything first. As the disk is being loaded you will receive something like the following display to keep you aware of progress:

```
                        |STRING C3         S  6%
loading sample:- STRING C2
```

***NOTE**: On hard disks, the loading may be so fast that you barely see the names flashing on the screen.*

It is possible that the chosen file(s) will occupy more memory space than is actually available, in which case the loading process will be halted and you will receive this prompt:

```
                        |STRING C3         S  6%
!! Insufficient waveform memory!!
```

Any files which have been completely loaded into memory prior to the prompt will remain in memory, however.

Even if a file exists in memory with the same name as a file on disk, the disk file will still be loaded and the file in memory will be overwritten.

The type of loads you can perform are extremely flexible and comprise:

ENTIRE VOLUME — This will load the entire contents of the disk (and other disks of the same volume when using floppy disks) into the memory (programs, samples, drum settings, effects, Qlists and operating system).

ALL PROGS+SAMPLES — As the name suggests, all programs and samples on the disk will be loaded into memory. Any other files will not be loaded (drum settings, effects files, Qlists, operating system, etc).

ALL PROGRAMS ONLY — Only programs (those files marked with a 'P' in the display) will be loaded.

ALL SAMPLES — All samples (files marked with an 'S' in the display) will be loaded.

CURSOR PROG+SAMPS — After selecting this parameter, move the cursor to a program file, and press CLR or GO. The selected program will be loaded, and then the S3200 will examine the program to see what samples are used by the program. These samples will then automatically be loaded.

CURSOR ITEM ONLY — After selecting this parameter, move the cursor to any file (program, sample, effects files, Qlists or drum settings), and press CLR or GO. The highlighted file will then be loaded into memory.

OPERATING SYSTEM — If the disk contains an operating system, you can load the system from disk.

LOADING S900/S950 SAMPLES AND PROGRAMS

No special command is provided for S900 samples. Inserting an S900 disk will give you this display:

```
S900 DISK ! use only for reading
```

Simply select the appropriate load type and proceed as above. The S3200 display will inform you when a sample for the S900 is being read, and after each S900 sample has been successfully read, an additional message, 'unscrambling S900 sample' will appear, as the S3200 converts S900 to S3200 format (12-bit to 16-bit).

AUTO LOADING FROM DISK

If you turn on the S3200 with a disk in the drive, the contents of the disk will be loaded into memory. If the disk contains a copy of the operating system and this operating system is the same or a higher version number than the ROM version, this will be loaded.

If an internal hard disk is fitted and set to SCSI ID 5 and the operating system is on the first volume, the operating system will be automatically loaded from this if it is the same or a higher version than the ROM version and no floppy disk is in the drive at power-on.

It is a good idea for you to make a copy of any Operating System disks for your S3200 and always turn on the S3200 with the latest version inserted in the drive (see below for details on saving operating system to disk).

SAVING TO DISK

REMEMBER! When you turn off the S3200, all samples, programs and drum settings are lost. Save your work to disk if you want to keep it for another session.

By pressing the SAVE key from the main DISK page, you can save your edited programs, samples and other files to disk. Make sure that you have enough unprotected formatted disks available or enough room on your hard disk before you save.

The process of saving to disk is much the same as loading from disk. There are two major differences, however:

1) Disk space is measured in blocks - not percentage free space. One formatted MF2DD disk contains 783 blocks, and one MF2HD disk contains 1583 blocks.

2) The unexpanded S3200 is capable of holding more data than will fit onto a single MF2DD or MF2HD disk. If you try to save an entire volume with many programs and samples, you will have to use more than one disk.

***NOTE**: You cannot save continuous samples across more than one floppy disk. For example, if you have a 24 second sample and try to save it, it cannot store part of that sample on one disk and part on another. If you are using long samples like this, we recommend you invest in a hard disk.*

When you first press **SAVE**, all programs, samples and drum settings in memory are displayed on screen, together with the amount of space in blocks that they will take up on disk.

```
SAVE TO DISK     : FLOPPYH vol: NOT NAMED
 free blocks:1399   |STRINGS 1       P   1
free entries: 115   |STRINGS 2       P   1
type of save:-      |SLOW STRINGS    P   1
 ENTIRE VOLUME      |STRING C2       S 345
progs:  3 samps:  7 |STRING C3       S 365
 LOAD SAVE REN  DEL  HDSK FORM WIPE  GO
```

Select the option to save: ENTIRE VOLUME, ALL PROGS+SAMPLES, ALL PROGRAMS ONLY, ALL SAMPLES, CURSOR PROG+SAMPS, CURSOR ITEM ONLY or OPERATING SYSTEM in the same way as for loading.

If you choose one of the CURSOR items, move the cursor to the sample or program in memory that you want to save. Saving CURSOR PROG+SAMPS will automatically save any samples associated with the highlighted program. If the samples already exist on disk with the same name, they will be overwritten. Be careful if you are using samples which have been slightly modified between programs; give them different names to avoid overwriting what may represent hours of work unless, of course, you specifically want to.

If you want to check the files already on disk, you can press **LOAD** to view files on disk and then return to **SAVE**.

When you have made your selection press **WIPE** or **GO** to save your work. **WIPE** will erase all data already on the disk, and save the selected file(s), and **GO** will simply save the files in addition to any already on disk. If you are saving to floppy disk, and there are more files to be saved than will fit on a single disk, you will be prompted to insert a new disk.

***NOTE**: It goes without saying that write protect must be off in order to save to a floppy disk!*

NOTES ON SAVING THE OPERATING SYSTEM TO DISK.

The S3200 contains its operating system on ROM. However, it is possible to upgrade software via floppy disk.

There are, however, benefits to loading the operating system from floppy in that you are able to save your own system defaults which will override those set at the factory. For example, if you have a particular way of recording and sampling, you can save all the record parameters such as default sampling time, start method, etc.. You may also save such things as digital input settings and hard disk SCSI ID's and sector size. When you load from floppy, these will be loaded.

You may even save your own test program by setting the parameters as you think fit and saving that as an ordinary program to the operating system disk. This too will be loaded on power up giving you maybe a more suitable template to work from.

Saving to hard disk follows the same procedure as saving to disk. Note that if the hard disk is divided into partitions, you can load data from one partition, select another partition on the SAVE page, and save it into the newly-selected partition. It is not possible to transfer data directly between partitions - it

must be done by loading it into memory and saving it back elsewhere on the disk.

> ***NOTE**: If you are using a Syquest removable cartridge or Magneto Optical drive, the write protect switch of the cartridge must be switched to OFF in order to save.*

RENAMING FILES

In the rename page, you can rename individual files on disk, or rename a volume on disk (a floppy disk can contain only one volume, but a hard disk can contain many volumes). If you have an internal hard disk fitted and/or and external drive attached, select the partition, and the volume to be renamed, or the volume containing the file(s) to be renamed, otherwise, if you're using floppy, insert the disk which contains data to be renamed. Press the REN key to enter a new name. You will receive this screen display:

```
RENAME ON DISK: FLOPPYH vol: NOT NAMED
new name:-             |STRINGS 1       P  0%
 NEW NAME              |STRINGS 2       P  0%
vol load number:       |SLOW STRINGS    P  0%
vol load enable:OFF    |STRING C2       S  5%
rename VOL or FILE     |STRING C3       S  6%
LOAD SAVE REN DEL HDSK FORM VOL FILE
```

or this if you are using a hard disk:

```
RENAME ON DISK: HARD-:A vol: VOLUME 021
new name:-             |STRINGS 1       P  0%
 NEW NAME              |STRINGS 2       P  0%
vol load number:OFF    |SLOW STRINGS    P  0%
vol load enable:OFF    |STRING C2       S  5%
rename VOL or FILE     |STRING C3       S  6%
LOAD SAVE REN DEL HDSK FORM VOL FILE
```

To enter the name, press the NAME key and type in a suitable name of up to 12 characters. You may enter numbers from the numeric keypad by pressing NAME again and you may toggle between the numeric keypad's letters or numbers simply by pressing the NAME key. You will get a screen display something like this:

```
RENAME ON DISK: FLOPPYH vol: NOT NAMED
new name:-             |STRINGS 1       P  0%
 NEW NAME              |STRINGS 2       P  0%
vol load number:       |SLOW STRINGS    P  0%
vol load enable:OFF    |STRING C2       S  5%
rename VOL or FILE     |STRING C3       S  6%
LETTERS .. (NAME for numbers ENT to exit)
```

When you have entered the new name, press ENT and then press VOL to rename the hard disk volume or floppy disk or highlight a file with the CURSOR keys and press FILE to rename the highlighted file.

With a hard disk fitted, MIDI Program Change messages may be used to load volumes. Use the `vol load number:` parameter on this page to assign a number from 1 to 128 for the current volume. Once set, you may turn this on and off freely in the `vol load enable:` field. On receipt of a Program Change message, the S3200 will scan all the hard disk volumes for a number set in this page which corresponds to the Program Change number in the MIDI

message. The volume will then be loaded (the currently-selected program number will change to 1 and program number 1 of the volume which has just been loaded will be selected).

> ***NOTE**: To rename a disk or a file on a floppy disk, removable cartridge or Magneto Optical disk, write-protection must be off.*

DELETING ITEMS FROM DISK

Pressing the DEL key displays this screen:

```
DELETE     disk : FLOPPYH vol: NOT NAMED
free blocks: 1399  |STRINGS 1        P  0%
type of delete:    |STRINGS 2        P  0%
 CURSOR ITEM ONLY  |SLOW STRINGS     P  0%
                   |STRING C2        S  5%
                   |STRING C3        S  6%
 LOAD SAVE REN  DEL HDSK FORM          GO
```

You can delete a file (or files) from a floppy disk or hard disk. If you have a hard disk fitted, select the volume using the parameter field on the first line. There are a number of options that you can pick to determine what file(s) will be deleted.

CURSOR ITEM ONLY	As its name suggests, deletes only that file which is highlighted by the cursor.
ALL PROGRAMS ONLY	This deletes all programs, but not their associated samples.
ALL SAMPLES	This deletes all samples on the current volume.
ENTIRE VOLUME	This is the most drastic, erasing all data on the volume.
OPERATING SYSTEM	This removes the operating system from the volume.

> ***NOTE**: In order to delete a file or files from a floppy disk, removable cartridge or Magneto Optical disk, write-protection must be off, of course.*

HARD DISK CONTROL

SCSI (Small Computer Serial Interface) has become very popular as a means of interfacing devices and as the S3200 is fitted with the IB-301S SCSI Interface board, you may use a wide range of hard disks as a storage device for your sound library. Hard disks offer more in the way of size and also speed. More recently, the removable cartridge types of storage devices have become increasingly popular. The problem with fixed drives is that when they fill up you either have to delete files (or back them up to floppy or DAT) or you need to buy another drive. With the removable types, you simply insert another cartridge. The problem with these, however, is that, whilst they are fine in daily use with a word processor, loading huge volumes of sound data into the S3200 all day in a busy studio can give rise to unreliability.

Perhaps the most elegant solution at the time of writing is the Sony Magneto Optical (MO) drive, especially the larger 650MByte type. This has all the benefits of a fixed hard drive in that storage capacity is very large and the

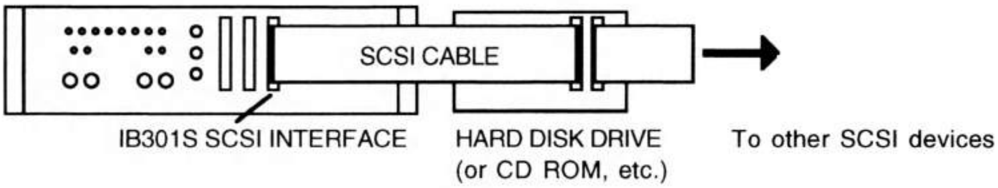
medium is also removable. They also seem to be extremely reliable. Although they can be expensive, they do come highly recommended. It is also possible to fit the SMO-P301 3.5 inch magneto optical drive into the S3200 which can be used almost as a giant 105Mbyte floppy drive!

The S3200 is compatible with all of these (*) including the two formats of MO disk - the 1kB/sector and 512B/sector.

** **NOTE**: New drives are appearing all the time and it is impossible for us to check every drive. Please be sure to really check with your dealer and/or Akai distributor that the drive you intend to purchase is able to work successfully with your S3200. Whilst every effort is made to ensure compatibility between drives, some devices to not adhere exactly to the SCSI protocol and can cause problems. Akai cannot accept responsibility for lost data*

CONNECTING AN EXTERNAL HARD DISK DRIVE

Connection to an external hard disk drive is as follows:



Take a standard 50 pin SCSI cable and connect it to the IB-301S connection and then to the SCSI connection on the hard disk drive. Most drives have two SCSI connections and the other can be used as a 'thru' to other SCSI devices such as CD ROM or another hard disk drive, etc..

NOTES ON USING HARD DISK DRIVES

• SCSI CABLES

Always high quality SCSI cables. Using low quality ones will give rise to data errors. The flat ribbon cables are really intended only for use inside devices and have insufficient screening which may cause data noises to appear in your audio signal path when any disk activity is going on, especially if your audio connections run parallel with the SCSI lead. However, if this is not a problem for you (and in a studio it may not be), these cables are usually very good. In a live situation, however, they are probably not ideal.

• TERMINATION

A chain of SCSI devices must be terminated at either end and terminating resistors are fitted to most SCSI devices when you buy them. The S3200 is terminated on the IB-301S SCSI interface board. It is most likely that the S3200 will be at one end of the chain and so must be left terminated. Any disks in between the S3200 and the last SCSI device in the chain must be unterminated (this can be done by physically removing the resistors or sometimes via a DIP switch on the back of the unit). The last device must be terminated. Incorrect termination may give rise to data errors so please ensure it is done correctly. If you have any doubts, please contact your dealer who will be able to help.

Your hard disk is controlled from the HDSK page of the DISK mode. When HDSK is pressed, the following screen display is seen:

```
HARD DISK CONTROL
                MO drive: ON
           SCSI drive ID: 5
           local SCSI ID: 6
  SCSI drive sector size: 512b
Press PARK to set heads to safe position
LOAD SAVE REN DEL HDSK FORM Sres PARK
```

The parameters on this page are used to set SCSI ID's. The most important parameters you need to know about are `SCSI drive ID:` and SCSI drive `sector size:`.

SCSI allows up to 8 devices to be on the buss and these each have a unique number 0 - 7 (it's similar to the concept of MIDI channels). In order for one SCSI device to 'talk' to a hard drive, their SCSI ID's must match. The `SCSI drive ID:` field sets the SCSI ID for the IB-301S interface. The default setting in the `SCSI drive ID:` field is 5 - this is the number given to an internal hard disk if one is fitted. If you use an external drive, this parameter must be set to match that of the drive. For example, if your external drive is 1, this parameter must be set to 1. You may, if you wish, have several drives connected and each of these must have different SCSI ID's. Each one can be selected by changing the ID number in the `SCSI drive ID:` field.

The `SCSI drive sector size:` field allows you to switch between the different MO formats of disk. There are two - 512Bytes per sector and 1kByte per sector. Both offer the same storage, they just have different sector sizes. The S3200 can use both and this is selected in this field. The default is `512b`.

The other field, `local SCSI ID:`, sets the S3200's SCSI ID (as distinct from the IB301's ID) and is used when communicating between samplers or computer editors over SCSI.

If you have the SONY SMO-P301 3.5 inch magneto optical drive fitted internally in the S3200, the `MO drive:` field allows you to turn this off in software. Three options are available. The first is `ON` which leaves the MO drive operating at all times. However, because the drive can be noisy due to the fans required for cooling, when not in use, you may like to turn it off. This can be done in two ways. Firtsly, you may set this parameter to AUTO which will turn the MO drive off as soon as you attempt to make a sample in the REC2 page of EDIT SAMPLE. In this way, the noise of the drive's fan will not spill over into your sample if you are using a microphone in the same room as your S3200. As soon as the sample has been recorded, the drive switches back on immediately. The other option is `OFF` which, as the function imples, turns the drive off completely. Please note that this does not apply to external MO drives that may be connected.

NOTE*: If your external drive's ID is something other than 5 and uses a lkBytes/sector disk, you may set the appropriate settings here and, by saving the operating system to floppy disk and powering up with it in the drive, the S3200 will default to the SCSI ID and sector size of your drive.*

PARK is a very important operation if you have a hard disk fitted and should be performed every time you end a session with a S3200 with a hard disk

fitted. If you do not have a hard disk fitted, this does not apply to you. The PARK procedure makes the hard disk safe for transportation. If you neglect to do this, you stand the chance of losing the data on the hard disk and the hard disk itself if the S3200 is roughly handled. To park the heads, press PARK. If, for some reason, the heads on the hard disk are not parked properly, a message will tell you to try again. If you keep re-trying and this message continues to appear, contact your AKAI dealer. Head parking is not a luxury, it is a necessity if you intend moving your S3200.

FORMATTING DISKS

Before you can use a disk, it must be formatted. MF2DD disks will automatically be formatted as low-density, and MF2HD disks will be formatted as high-density. As explained earlier in this manual, these different types of disk cannot be formatted in the other way as they could on the S1000 and S1100.

FORMATTING A FLOPPY DISK

To format a floppy disk, insert the disk in the drive, and press FORM. You will see this screen display:

```
FORMAT FLOPPY OR HARD DISK : FLOPPY
                 BLOCKS     HARD PARTITIONS
track:         good:         size: 60 Mb
 side:          bad:

FORMat or ARRange floppy disk:-> rSTART¬
LOAD SAVE REN DEL HDSK FORM FORM ARR
```

Select FLOPPY at the top of the screen if it isn't already selected and press FORM. You will see:

```
formatting disk HIGH DENSITY. . .
```

or

```
formatting disk LOW DENSITY. . .
```

depending on the type of disk you are using. The process will take about a minute and the track and side number of the disk will be displayed as the operation proceeds. When the operation is complete you should receive the following display:

```
FORMAT FLOPPY OR HARD DISK : FLOPPY
                 BLOCKS     HARD PARTITIONS
track:    80  good: 1583     size: 60 Mb
 side:     2   bad:    0
     DISK IS READY FOR USE
FORMat or ARRange floppy disk:-> rSTART¬
LOAD SAVE REN DEL HDSK FORM FORM ARR
```

This indicates that the disk has formatted correctly and is safe to use. If you get an indication that there are bad blocks, the disk may be unreliable. The S3200 will inform you of this. If you do receive such a message, you may like to try again but it usually means that the disk has become seriously damaged in some way. This is unlikely to happen on brand new disks but may happen on floppy disks that are being re-used, especially if they have been used before on another system.

Formatting a disk will permanently remove all data previously recorded on it. Only format new disks or ones which contain data that you are sure you don't need any more.
If you have a hard disk connected to the S3200 as well, be especially careful to select FLOPPY!!

The ARR key is used as a quick format for changing the directory size of disks formatted on an S1000 or S1100 (please see below - NOTES ON USING S1000 AND S1100 SOUND LIBRARY).

Usually, you must use FORM to format a new disk for use in the S3200. Trying ARR on an unformatted disk will display the prompt:

```
can't quick-format this disk !
```

You should use FORM.

FORMATTING A HARD DISK

Hard disks can and should also be formatted before use. The maximum size of hard disk which can be formatted and used with the S3200 is 510Mbytes. If any larger hard disk is attached to the S3200, data above this size will not be recognized or used. Switching to HARD-: will display this screen:

```
FORMAT FLOPPY OR HARD DISK : HARD-:
                  BLOCKS      HARD PARTITIONS
part.:            good:         size:60 Mb
 size:             bad:          max: 2

FORMat or ARRange  hard  disk:-> rSTART
LOAD SAVE REN DEL HDSK FORM FORM ARR
```

For convenience, large hard disks are split into partitions, which are named A, B, C, etc (if you are used to MS-DOS systems, these partitions are analogous to 'logical drives' on a hard disk). All partitions must be the same size, which you can select with the HARD PARTITIONS size: parameter to be variable between 1 and 60Mbytes. The last partition on a hard disk takes up all the remaining space on the disk (i.e. on a 120Mbyte disk divided into 50Mb partitions, A and B will both be 50Mbytes, and C will be 20Mbytes).

A further field on this page is the max: field. This allows you to set the number of partitions you wish to create and this is included for the disk recording functions. For example, if you have 300Mb hard disk, you may set it to have 4 x 50 megabyte partitions by setting 50 and 4 respectively in the size: and max: fields. This would leave 100Mb free for disk recording giving you 10 minutes of stereo recording at 44.1kHz. In this way, one disk may hold not only sound library but disk recordings as well. This is explained later in this manual in the section that explains the disk recording functions. If you intend to do this, it is best to check this before formatting your hard disk - formatting it later in order to use the disk recording functions will erase any sound library you may have already saved.

To format the hard disk, press either FORM or ARR depending on the action you want to take. You will receive the following safeguard prompt:

```
FORMat or ARRange hard disk:->> rSTART
 DESTROY ALL HARD DISK DATA ??    NO  YES
```

Answer NO if you have second thoughts, otherwise answer YES.

Both formatting and arranging will destroy all data on the hard disk.

Arranging is a faster operation than a full format (it simply initializes directories into a format suitable for use by the S3200). Make sure that there is no data stored on the hard disk which you want to keep.

Formatting will take a few minutes, followed by the arrange process. Bad blocks will be automatically 'swapped out' in a verification procedure. You can bypass this verification process by pressing SKIP, but it is suggested that you let it run its course - it will end up safer in the long run.

If the drive is not connected or the SCSI ID's don't match, you might receive the message when you press FORM or ARR:

```
waiting for hard disk ready. .  SKIP
```

or it may say:

```
HARD DISK DRIVE NOT READY !
```

Please check your SCSI cables and that the drive is switched on (it does happen!). Also, please check the SCSI ID settings of both the drive and the IB-301S. You will also get this message if a removable type of hard disk is being used and the disk is not inserted in the drive regardless of connections and settings.

NOTES ON USING EXISTING S1000/S1100 SOUND LIBRARY

To accommodate the need for larger disk directories, the S3200 now allows 512 items to be saved on a floppy or hard disk (previously it was 64 for floppy and 128 for hard disk). As a result, the whole format of the directories is completely different.

S1000 and S1100 disks can, of course, be used with no problem (*). The problem occurs, however, if you try to save to a disk that was formatted on an S1000 or S1100. Because the disk directory has now changed, the S1000/1100 disk has to be reformatted. When you perform a save, if you use WIPE, this process is done automatically for you. If, however, you use GO instead of wipe, the S3200 will remind you:

```
re-format or arrange before writing !
```

You may either specifically go through the formatting procedure yourself or you may simply press WIPE.

The same is true of hard disk volumes. Simply using the GO key when you try to save to a hard disk volume that was originally formatted using an S1000 or S1100, will cause this message to be displayed:

```
Must kill S1000 volume before writing !
```

This is saying that this volume's directory needs to be re-written in the S3200 format. To do this, you should use WIPE - this will automatically rewrite the directory. After this, you may use the volumes as normal.

WARNING!

MAKE SURE ALL DATA IS EITHER SAVED ELSEWHERE OR STORED IN THE SAMPLER. REWRITING THE DIRECTORY WILL ERASE ALL THE HEADERS FOR THE SAMPLES, PROGRAMS, EFFECTS FILES, ETC., AND YOU WILL LOSE YOUR DATA.

WHEN USING A HARD DISK, THE ACTION OF KILLING A VOLUME APPLIES ONLY TO THAT VOLUME. THIS IS NOT A FORMATTING PROCEDURE. WHEN YOU SEE THE PROMPT, USE WIPE - DO NOT FORMAT YOUR HARD DISK AS THIS WILL ERASE EVERYTHING ON IT.

***(*) IMPORTANT NOTE**: On the S1000 and S1100, it is possible to format DD disks to a high density format. This is not possible on the S3200. Furthermore, the S3200 cannot read DD disks that have been formatted on an S1000/1100 to a high density format. You will need to first load these sounds into an S1000/1100, resave them onto high density disks (formatted to high density, of course) before they can be used in the S3200. High density disks have a hole on the right hand side which is used by the disk drive to detect that it is a high density disk. If the S3200 does not 'see' this hole, it assumes it is a DD disk and so searches for a DD format. If it doesn't find it (i.e. because the disk is high density format), it cannot read it.*

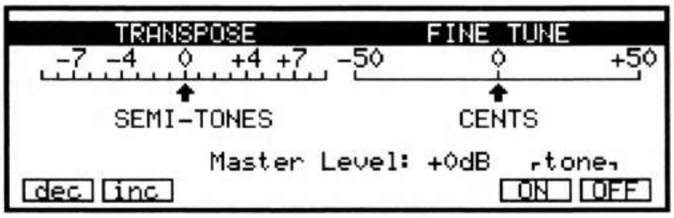
PARAMETER SETTINGS

If you are very familiar with your programs or have an S1000/S1100 to directly compare with, you will notice that some parameter values are different in the S3200. This is not a fault but a 'fix up' our software engineers have done so that S1000 or S1100 sounds loaded into the S3200 sound the same.

On the S3200, many of the program parameters (and some EDIT SAMPLE parameters) have different ranges and so some offsets are invoked so that, for example, LFO speeds are consistent between the two families of samplers.

TUNE/LEVEL SCREEN

Pressing the TUNE/LEVEL mode select key gives you this screen display:



TUNING AND TRANSPOSING

The S3200 can be transposed by up to ±9 semitones and fine tuned by up to ±50 cents (one semitone) to enable easy playing in difficult keys and to match tuning with other instruments. When you first press the TUNE/LEVEL key, two scales indicate the current transposition and tuning. Use the CURSOR < and > keys to transpose up or down and the DATA control to provide fine tuning (one click of the knob equals one cent). These transposition and tuning settings will be lost when power is turned off unless they are saved to disk in a full volume save.

There are two soft keys ON and OFF in this screen display on F7 and F8. These will turn an A=440Hz audio signal on and off to the stereo output connectors (and the headphones). This may be used as a tuning reference for the sampler (or any other instruments you have) or as a test tone for checking levels, etc..

SETTING THE S3200'S MASTER OUTPUT LEVEL

As well as the main volume control, it is also possible to set the master level for the S3200 in this page. The primary benefit of this function is to set the output level to match different mixers' headroom. It is possible to boost the sampler's output level for a 'hotter' output for professional +4dBm desks but for desks that run at -10dBm, you may prefer to cut the level back a bit to prevent distortion. In order to optimise the S3200's signal to noise ratio, it is recommended you run the outputs as high as possible - this, in turn, will require less gain on your mixer input channels which will keep noise levels down. The level settings will be lost when power is turned off unless they are saved to disk in a full volume save.

Level is adjusted using the dec. inc. soft keys - F1 and F2.

You will note that this control affects not only the stereo outputs but also the individual outputs.

NOTE 1: *Adjusting the master output level will affect the level of the audio appearing at the real time output of the digital audio interface board as well.*

NOTE 2: *All of the tune and level parameters are saved to disk when a full volume save is performed. They are not saved with any other type of save. This is also true of loading from disk - a full volume load will load these parameters but any other type of load will not.*

UTILITY MODE

The UTILITY mode is one where several ancillary functions are included. Here you may remotely program the Akai ME-35T drum-MIDI converter as well as access the Qlist facilities and the digital back-up functions. You also access the S3200's disk recording functions in the UTILITY mode. On entering this mode, you will see this screen:

```
UTILITIES
The following utilities are available:
* ME35T Interface
* Digital Audio
* Cue-lists (smpte)
* Direct-to-disk Recording
DRUM DIGI SMPT  DD
```

No parameters are directly available here - this merely displays the options open to you.

PROGRAMMING THE AKAI ME35T

Pressing DRUM displays the following screen:

```
DRUM INPUT SETTINGS  name: DRUM INPUTS
unit:  1      input: ALL
chan:  1    capture:  4mS
note: 60    recover: 10mS
sens: 50    on-time: 10mS
trig: 25    V-curve:  3    IN:-1 2 3 4 5 6 7 8
 EDIT CONT
```

The S3200 is capable of acting as a highly sophisticated percussion sampler using the AKAI ME35T audio/MIDI trigger interface unit to produce MIDI trigger signals from a variety of sources. Two such units may be connected, and programming of them may be carried out from the S3200 rather than on the more limited displays and controls of the ME35Ts. For such programming to take place, a 'MIDI handshake' must be set up, from IN to OUT and OUT to IN.

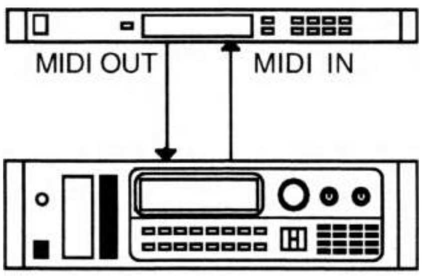


Fig 1

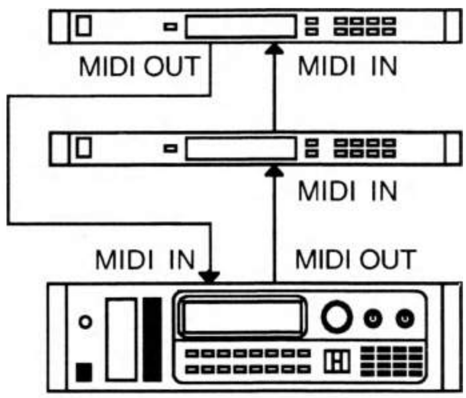


Fig 2

When two ME35T units are to be used together to provide 16 drum inputs, they should be connected as shown in Fig 2.

Since this is a manual for the S3200, full details of the operation of the ME35T will not be given here. Refer to the ME35T manual for operational details. However, note that to set up the MIDI Exclusive channel on the ME35T, the MIDI CHAN and MIDI NOTE keys on the ME35T should be pressed simultaneously. The following parameters on the ME35T may be set up from the S3200:

The name of the drum input settings may be altered by pressing NAME, typing in the name followed by ENTER.

The parameters on this page are as follows:

unit:	Either one of two ME35Ts may be selected for parameter editing here.
input:	Selecting ALL allows the inputs to be globally edited to rough values, and then individual (1-8) inputs may be selected for fine adjustment. This method of working can save you a lot of time.
chan:	Here you may select the MIDI channel for the selected input.
note:	Here you may select the MIDI note number you wish to assign to the input
trig:	This sets the trigger sensitivity of the selected input and should be adjusted to match your playing style and also to the nature of the drum pad, mic or bug you are using. As you adjust the trigger level, this is represented in the box to the right.
capture:	This allows you to set the capture time of the selected input.
recover:	This allows you to set the recovery time of the selected input and should be set so that stick bounce doesn't cause unnecessary false triggering.
on-time:	This sets the length of the note that will be issued from the ME35T's MIDI output for that channel. In this way, drums can be used to trigger keyboard sounds.
V-curve:	Here you may select from 8 different velocity curves to match your playing style. Please see the ME35T manual for details of these curves.

As you play your pads or drums, you will see something like the following display with a PPM style bargraph:

```
DRUM INPUT SETTINGS   name: DRUM INPUTS
unit:  1      input: ALL
chan:  1    capture:  4mS
note: 60    recover: 10mS
sens: 50    on-time: 10mS
trig: 25    V-curve:  3     IN:-1 2 3 4 5 6 7 8
 EDIT CONT
```

For more information on these parameters, please refer to the ME35T's operators manual.

The second page of the DRUM mode is accessed by pressing CONT:

```
DRUM UNIT CONTROL
                        UNIT 1      UNIT 2
           operation:    ON          OFF
   exclusive channel:     1            2
    MIDI thru enable:    OFF         OFF

EDIT CONT
```

This page allows you to set up MIDI parameters for up to two ME35T units. Parameters which you can set are: operation: (ON or OFF), exclusive channel: for programming (1-32) and enable of MIDI THRU operation (ON or OFF). To return to the first DRUM page, press EDIT. You may exit the DRUM mode by pressing the UTILITY mode select key again - this will return you to the main UTILITY page.

DIGITAL AUDIO INTERFACE

The third option in the UTILITY mode is the DAT BACKUP facility. This uses the IB-302D AES/EBU Digital Audio Interface supplied with the S3200.

REAL TIME DIGITAL OUTPUTS

The IB-302D is always transmitting digital audio - this is a digital copy of the audio appearing at the LEFT/RIGHT stereo analogue outputs. This can be sent directly to other digital audio devices that can accept AES/EBU and SPDIF digital audio signals such as DAT, a digital multi-track recorder such as the Akai ADAM DR1200 (using the DIF1200) or Akai DD1000 Magneto Optical Disk recorder/editor. No special setup is required for this.

DAT BACKUP

One very useful function of the digital interface is that of DAT backup. This allows you to make safety copies of your data on a simple DAT tape. This can be invaluable for archiving a disk when it is full and helps you overcome the problem of lost data in the event of a problem occurring with your hard disk. To access the DAT BACKUP functions, press F2 - DIGI. You will receive this screen display:

```
DIGITAL BACKUP                          programs:
  current vol: NOT NAMED                 samples:
complete vols:                            Qlists:
  backup type: PARTITION:A                Tlists:
     transmit: 44.1kHz CONSUMER             Fix:
                                           drum:
      DIGI                          SAVE  LOAD
```

Here you may set the parameters and perform a backup or restore. The fields are as follows:

`current vol:` This displays the name of the current volume selected in the DISK mode. When the backup or restore process is in operation, this field changes to show the name of the volume currently being backed up or restored. This field is not accessible to the user.

`complete vols:` This field is also not accessible to the user and merely a progress display to show the number of volumes that have been backed up or restored in the process.

`backup type:` Here you may select to backup either one partition of your hard disk or just a single memory load of samples - i.e. those in RAM at the present time. Usually the selection is `PARTITION:` because you mostly want to use this facility to make safety copies of your hard disk. In this field, you may move the cursor to where it says 'A' and select the partition you want to backup. If you select `MEMORY`, you may back up or restore the current contents of the S3200's RAM. Many people, especially those with memory expansion boards fitted in their sampler, use the memory back up facility to backup long samples rather than tie up an expensive hard disk drive.

transmit: The sampler's transmit rate is fixed at 44.1kHz. Here you may select between the consumer format for digital audio (SPDIF) or the professional AES/EBU format when transferring digital audio to DAT or some other recording medium or when performing DAT backup. What you select depends very much on your equipment. Some equipment is very forgiving and doesn't mind either format. Other equipment, however, is not so flexible and you have to choose specifically which format to use.

PERFORMING A DAT BACK UP

This is simple! Assuming you have made the necessary digital audio connections, simply press RECORD and PLAY on the DAT (ar whatever combination is required on your particular machine) and, ensuring that the DAT is actually recording, press SAVE - F7. The S3200 will backup to DAT.

If you are saving volumes to DAT individually, 2 Meg will take about 30 seconds. If you are backing up the contents of your hard disk to DAT, each 2 Meg volume takes about 100 seconds - this is due to certain SCSI control considerations.

Data is saved in a digital audio format and but the sample headers and program information and other data are stored in a special format. During the save, both DAT channels are used but when restoring, only the left hand side is used.

PERFORMING A DAT RESTORE

This too is simple. Find the point on the DAT where your backup is, press LOAD on the S3200 and PLAY on your DAT - the contents of the S3200's memory or hard disk will be restored.

Please note that when performing a restore, it is not possible to do a partial restore - i.e. when restoring into memory, you cannot selectively select one snare drum, for example, and when restoring to hard disk, you cannot selectively choose a single volume.

As backing up and restoration takes place, the number of samples, programs, Qlists and other items will be displayed on the screen.

NOTE: *The DAT backup/restore process is only really capable of reliably doing this when using the same drive. For example, it cannot be guaranteed that backing up to an MO drive and then trying to restore it to a Syquest cartridge or fixed hard disk will work successfully. This is not a fault of the S3200 but due to differences in disk speed. Having said that, you may be successful trnasferring from one type of disk to another.*

PLEASE NOTE
THE DAT RESTORE PROCESS WILL ERASE THE CURRENT MEMORY CONTENTS OR HARD DISK DATA - PLEASE BE CAREFUL

NOTE*: Though digital tape streaming offers many advantages in terms of speed, convenience and cost of the media, Akai does not guarantee the integrity of the data backed up in this way. It is therefore strongly recommended that for archive purposes you also use floppy disks.*

SMPTE PAGE - USING CUE LISTS

Pressing SMPT in the main UTILITY page will take you to the Qlist mode where you may set up Qlists for synchronising to timecode amongst other things.

One popular application for samplers is to use them to dub sound effects in audio/visual productions. In the past, the only way to trigger these sound effects was to use a MIDI keyboard or sequencer and most audio/visual post-production suites do not employ MIDI but use timecode to synchronise equipment and trigger sound effects. Using the IB-303T SMPTE reader/generator fitted in the S3200, you may create sophisticated cue lists making it ideal for audio/visual post production. Pressing SMPT takes you to the cue list PLAY screen. This should show an empty cue list if you have entered this mode for the first time. I.e:

This is the QPLAY mode where you may play, create and edit cue lists. Only one cue list can reside in the S3200 at any one time and the maximum number of events a cue list can have is 250.

The basic premise is that you specify a SMPTE time for an event and this sends a MIDI note to one (or more) of the S3200's internal programs. Any MIDI note may be specified at any MIDI velocity to trigger any sound in any program in the S3200. Unlike other modes of the S3200, it is not necessary to specify program numbers for each of the programs as the S3200's cue list simply 'looks' at the program specified for the event and plays the appropriate note in that program. In other words, if you set the following for an event:

```
00:00:12:00.0   GUNSHOTS   Mon   C_3   127
```

the S3200 will send a MIDI note ON event (Mon)to play a gunshot sound effect on C3 in the GUNSHOT program at 12 seconds with a MIDI velocity of 127. You may have any number of programs in one cue list and these programs can contain any number of samples in keygroups on specific note numbers. You may layer two or more samples on top of each other within any one keygroup and these will be played simultaneously when the appropriate MIDI note number is sent from the cue list.

You may use the cue list in several ways. You could set up a program that contains all the sound effects for your production and assign these to different MIDI notes in their respective keygroups. In the cue list, you would simply assign the one program to every event and then specify the appropriate MIDI note numbers for each event. Alternatively, you could create a number of programs that contain just one sample in each program and these could be triggered by the cue list. Another method is a mixture of both techniques. You could have several programs, each with sound effects assigned to different MIDI note numbers. For example, you could put all your gunshot sound effects in program, all your footsteps in another, all your traffic noises in another

and set the cue list to play the appropriate sound effect (i.e the MIDI note number) from the required program.

As you have already discovered in EDIT SAMPLE, there are many different ways you can play back a sample in the S3200. A sound can be looped or it can simply play to the end of the sample. If the sound has no looping set in it, then it is not necessary to set a MIDI note OFF command to it when it is triggered in the cue list and so all that is required is a MIDI note ON event for such samples. If, however, the sample has a loop set in it, then it will be necessary to sent a MIDI note OFF in order to stop it sounding at the required moment although you will note that if the cue list is stopped during playback of a looped sample, the sample will automatically be stopped and you will not be left with a 'hanging' drone.

The 32 voice polyphony of the S3200 effectively gives you 32 tracks of audio available through the stereo outputs and/or the 8 individual outputs. Of course, you may also assign effects to the sounds in the usual way.

Before we look at the different modes within the cue list pages, let us first examine some of the functions of the cue list as these are very important in the creation and editing of cue lists.

BLOCKS

All editing of the cue list is done using 'blocks'. A block can be just one single event or a group of events and there are dedicated soft keys for identifying the start and end of a block which work in conjunction with the numeric keypad which functions as a cursor control in the cue list mode. When you identify a block, a highlighted box appears to the left of the selected event(s).

THE CURSOR

The cursor can be moved around the cue list in the usual way using the CURSOR keys but there is also a highlighted vertical field running down the side of the cue list called the 'scroll bar' and there is a pointer cursor that indicates the current position of the cursor in the cue list. This pointer cursor can be placed above or below an event and this is used to mark events for copying, insertion or deletion. You may also use this cursor to play the cue list from any point. In the cue list edit mode, there is an indicator at the top of the screen to show the current event number the cursor is placed on.

GETTING AROUND THE CUE LIST - THE NUMERIC KEYPAD

When you are in the cue list, the numeric keypad has a slightly different function to the other modes in the S3200 as it allows you to move the cursor up and down the cue list scroll bar.

Pressing 0 always takes you to the start of the cue list.

Pressing +/< takes you to the end of the cue list.

Pressing any of the number keys moves you down the list by the same number of steps as the number key you pressed. In other words, pressing 1 moves you down the list one event at a time, pressing 2 moves you two steps at a time, pressing 3 takes you three steps at a time, etc..
By pressing the -/> key AND a number key simultaneously, you can move up the list. For example, pressing -/> and 6 will take you six steps backwards in the list.

Pressing the +/< key AND the -/> key simultaneously takes you to the start of the block.

If you are on a field in the cue list (i.e. the cursor is not in the scroll bar), you may move to the scroll bar instantly by pressing the ENT key. When the cursor is on the scroll bar and pointing to an event, you may press the ENT key to play that event.

If the cursor is on a numeric field in the cue list, the numeric keypad functions in the normal way and this cursor control facility only ever works when the pointer cursor is on the scroll bar.

CUE LIST MODES

There are three basic modes in UTILITY for playing, creating, editing and setting up cue lists which are accessible through soft keys F1, F2 and F3. These are:

PLAY	This is where you play a cue list and this mode has several 'transport' controls for a variety of different playback functions.
EDIT	This is where you can edit and/or create a cue list.
SMPT	This where you set the S3200 SMPTE receive/transmit parameters.

CREATING AND EDITING CUE LISTS

Let us now create a cue list so press EDIT (F2) to take you to the cue list edit page:

```
 1 mt:+00:00:00:00.0      sl:+00:00:00:00.0
 00:00:04:21.5 DOOR SLAMS1      Mon C_3 127
 00:00:12:12.4 FOOTSTEPS        Mon C#3 127
 00:00:13:10.6 BG MUSIC 1       Ton  50 MID
 00:00:14:05.1 BG MUSIC 1       Tof  50 MID
 00:00:14.16.7 GUNSHOT 1        Mon C_3 127
PLAY EDIT MARK BLCK INS DEL SLIP SORT
```

This shows a typical cue list with sound effects.

The fields across the top of the screen are:

To the far left of the screen is a highlighted numeric field that is inaccessible to the user. This shows the current event number and, as the cursor is moved up and down the scroll bar, this number changes to indicate the event number in the list.

mt:+00:00:00:00.0	This sets a master offset time for the cue list and data is input using the data wheel or the numeric keypad. The whole cue list can be offset forwards by pressing the +/< key when the cursor is in this field or backwards by pressing the -/> key when the cursor is in this field.

sl:+00:00:00:00.0	This allows you to set the time by which you want to slip a block of events in time and data is input using the data wheel or the numeric keypad. You may slip a block of events forwards using the +/< key when the cursor is in this field or backwards using the -/> key when the cursor is in this field. This facility works in conjunction with the SLIP soft key described below.

Before we look at the other fields on this screen, let us first examine the function of the soft keys as these play an important part in the creation of a cue list.

MARK	Pressing this key sets the start mark of a block as set by the position of the pointer cursor in the scroll bar.
BLCK	Pressing this soft key sets the end mark of a block as set by the position of the pointer cursor in the scroll bar.

To mark a block, move the cursor to the event you wish to set and press mark. Now move the cursor down the list using the numeric keypad as described earlier and press BLCK at the point you wish to mark as the end. It is possible just to mark one event and this is done by pressing only MARK at the event you wish to set and not pressing BLCK.

INS	This allows you to copy and insert a marked event or block. To insert an event, place the pointer cursor at the event you wish to copy and press MARK. Now move the cursor using the numeric keypad to the position at which you wish to insert this event and press F5 - INS. The marked event will be inserted at the point indicated by the pointer cursor. You may now edit this event if you wish.

You may also copy and insert blocks in the same way. Place the cursor at the point at which you wish to set the start of the block and press MARK. Now, using the numeric keypad, move the cursor to the point you wish to set as the end of the block and press BLCK. You may now move the cursor to the position at which you wish to insert this block and press INS. The marked block will be inserted at the point indicated by the pointer cursor. You may now edit the cues in that block if you wish.

DEL	This deletes the marked event(s)
SLIP	This allows you to move a marked event or block of events backwards or forwards in time. This is most useful on blocks of cues that need to be shifted very slightly. When you press SLIP, the cursor will automatically be placed on the 'sl:' (slip time) field at the top right of the screen and you may set this field accordingly. After that, pressing SLIP - F7 - will slip the marked event(s) by the amount set in the 'sl:' field.
SORT	This allows you to sort a cue list's events into chronological order. The S3200 will play events in the right sequence even if they appear out of time on the screen but, to make things clearer for yourself, you

may want to place them in strict time order so that they follow consecutively on the screen display.

When you first enter the EDIT CUE page, you will always be presented with one cue. This is a MIDI note ON cue (Mon) set at 00:00:00:00.0 with no program assigned. I.e.:

```
 1 mt:+00:00:00:00.0        sl:+00:00:00:00.0
   00:00:00:00.0                 ? Mon 50 MID



PLAY EDIT MARK BLCK INS DEL SLIP SORT
```

The default MIDI note is C3 (note number 60) and the default velocity setting is 127. You may use this as the basis for your cue list using the various copy and insert functions. To create a number of cues, simply press F5 - INS - a few times to create some new cues. Now, simply set the appropriate SMPTE times and assign the relevant program(s) and MIDI note number(s). You may also specify the velocity setting for the cue(s) if you wish to affect a cues loudness or volume. You will note that a SMPTE time other than 00:00:00:00.0 HAS to be specified for any cue as this SMPTE time effectively represents 'no time' and so the cue will not sound.

If the sound has no looping (i.e. in EDIT SAMPLE, it is either set to "PLAY TO SAMPLE END" or no loop has been set) then it is not necessary to specify a MIDI note OFF for every event - only when a sample has a loop set in it is it necessary to specify a MIDI note OFF or, if a long sample needs to be shut off before it reaches the end of its duration. This is done by changing Mon to Mof. If you wish, you may specify a note off command (Mof) at any time to cut a sample short without having to edit its end point in EDIT SAMPLE.

You may also select takes to play from disk. This is done simply by changing Mon to Ton (TAKE ON). This is all explained more fully in the section that deals with using disk recordings in Qlists. Please refer to that section for more details.

You may now copy and insert either single events or blocks of events as you wish until the cue list is complete. At any time, to hear the results of your efforts, press F1 - PLAY - to take you to the QPLAY screen. You will receive this display:

```
QPLAY  QL1              time:   :  :  :  .
  00:00:04:21.5 DOOR SLAMS1      Mon C_3 127
  00:00:12:12.4 FOOTSTEPS        Mon C#3 127
  00:00:13:10.6 BG MUSIC 1       Ton 50  MID
  00:00:14:05.1 FOOTSTEPS        Mon C#3 127
  00:00:14.16.7 GUNSHOT 1        Mon C_3 127
PLAY EDIT SMPT GRAB Pext Pint Cint STOP
```

This page allows you to play the cue list from any point and there are several 'transport' controls that do this.

Pext This is an abbreviation for PLAY EXTERNAL and plays the cue list but only when it is receiving incoming SMPTE from an external source. If the external source stops, then the S3200 will stop. If the external source is 'rewound' to another location, the S3200 will pick

up on the new SMPTE position and re-commence playback from that point although you will note that if playback re-commences at a point that is halfway through a sample, that sample will not play. If no SMPTE is present, then nothing will happen when Pext is pressed!! When the cue list is playing, the cue list will scroll through the cue list and a small highlight appears to the left of the cues to indicate that the event has played. The current cue playing is always the middle cue in the screen except when playing the first three cues at the start of a cue list.

Pint

This is an abbreviation for PLAY INTERNAL and will play the cue list from its own internal SMPTE generator. It will also transmit SMPTE timecode through the SMPTE out socket on the rear of the S3200 allowing you to control external devices using the S3200 as the master controller. When the cue list is playing, the cue list will scroll through the cue list and a small highlight appears to the left of the cue to indicate that the event has played. The current cue playing is always the middle cue in the screen except when playing the first three cues at the start of a cue list.

Cint

This is an abbreviation of CONTINUE INTERNAL and pressing this after pressing STOP (see below) will play the cue list from the current position of the pointer cursor. This also transmits SMPTE timecode through the SMPTE out socket from the point at which playback is re-commenced. You can freely move the pointer cursor to any location in the cue list to commence playback from any point using this key. When the cue list is playing, the cue list will scroll through the cue list and a small highlight appears to the left of the cues to indicate that the event has played. The current cue playing is always the middle cue in the screen except when playing the first three cues at the start of a cue list.

STOP

Not surprisingly, this stops playback of the cue list in all play modes. It also stops transmission of timecode from the SMPTE out socket.

NAMING CUE LISTS

In this page, it is also possible to name a cue list. To do this, press NAME and type in the name (up to 12 characters) followed by pressing ENTER. The 'time:' field to the right of the name display shows the current time of either the internal or external timecode.

GRABBING TIMES

One other soft key that is available for use in the QPLAY mode is the GRAB function found on soft key F4. This allows you to input cues in real time as the cue list is playing. To do this, press either Pext (F5) or Pint (F6) and, as the cue list is playing, press GRAB at appropriate moments. This will input empty cues at the end of the cue list. You may edit these and assign the

relevant programs, MIDI notes and velocity levels in EDIT and pressing SORT in cue list EDIT will place them in their correct chronological order. This method of inputting cues is well suited for creating cue lists 'on the fly' - that is, watching the visuals whilst entering cues in real-time. You may use this function to add cues to an existing cue list or to create a cue list 'from scratch'.

Another way to do this is to use the numeric keypad whilst the cue list is playing. This will insert programs 1 to 9 in realtime as the cue list plays according to the key press and you will hear the sound as you do this. Naturally, it is important that your programs are numbered correctly 1-9 if you are to achieve the right results using this facility. Also, if no programs are assigned to any keys that are pressed, no input will be made.

The keys will normally insert the appropriate program on MIDI note C3 with a velocity of 127 but this may be changed by changing the parameters set in the TRANS page of the main MIDI mode (please refer to the section that explains the MIDI mode for more details on this function).

For either way of inputting cues in real-time, the GRAB soft key has two functions. If you press GRAB whilst the S3200 is NOT playing, a small 'G' appears at the top of the screen next to the 'time' field. When this is displayed and any of the PLAY functions are used, you can input cues in real-time (using the GRAB function or by 'playing' the programs from the keypad) but the cues will not be displayed as you input them. This allows a far faster response time for 'grabbed' cues. You may still input cues in real-time using either method without pressing GRAB first and you will see the cues entered as you 'play' them but please note that the response time is slightly slower because, as the screen display changes, so some of the S3200's execution speed is used up and it is possible in such a situation, especially when inputting really fast cues, that some cues may be missed. It is recommended, therefore, that you press GRAB before putting the S3200 into play if you need to input very fast cues in real-time. If you wish to cancel the 'G', press STOP (F8) and this will put you back to the normal GRAB mode.

You may not access any other fields in this mode as these are for display only although the CURSOR keys and the numeric keypad can be used to scroll through the cue list. If you wish to edit the cue list, press F2 to return to the EDIT screen.

SMPTE PAGE

This page is where you set the parameters for the S3200's internal SMPTE reader/generator. Pressing SMPT will give you this display:

```
SMPTE                                H  M  S  F  f/s
          receive time:-              :  :  :

        transmit start:-           00:00:00:00.0  25
      current transmit:-              :  :  :
                                       ┌tran┐
 PLAY EDIT SMPT                   RCUE STRT CONT STOP
```

`receive time:` This field shows the current time being fed to the SMPTE reader from an external source. It will also detect the frame rate used on the external source and this is displayed underneath the 'f/s' field shown in the top right hand corner of the screen.

`transmit start:` This field allows you to set the time at which you want the SMPTE time to start. To the right of this field is another that allows you to set the frame rate for the transmitted SMPTE and the options are 24 fps, 25 fps, 30 fps and 30 drop fps. It is important that this be set to match incoming external timecode otherwise you may find that certain cues 'misfire'.

`current transmit:` This field shows the SMPTE time currently being sent.

There are four soft keys associated with the reception and transmission of SMPTE. These are:

RCUE This switches the S3200's SMPTE reader/generator to receive external SMPTE/EBU timecode. When this is switched on and external timecode is sent to the S3200, the 'receive time:' field shows the current external timecode position and the 'f/s' field shows the the external timecode's frame rate.

STRT This generates SMPTE timecode from the S3200's internal SMPTE generator from the point set in the 'transmit start:' field.

CONT This generates SMPTE from the point at which the timecode transmission was stopped.

STOP This stops transmission of SMPTE from the S3200's internal SMPTE generator.

SAVING CUE LISTS

You may save a cue list and its programs and samples by selecting ENTIRE VOLUME as the save type - this will save all the programs, samples and effects file associated with the cue list to disk.

It is also possible to save any number of cue lists to disk. To do this, go to the DISK page and select CURSOR ITEM ONLY. Place the cursor on the cue list file you wish to save press F2 - SAVE - and then press F8 - GO. This will save the

cue list to disk and the suffix 'Q' will be shown alongside the file. Any number of cue lists can be saved to a disk although only one can exist in the S3200 at any one time.

LOADING CUE LISTS

When a disk is inserted into the S3200's disk drive on power up, the cue list file will be loaded along with the the programs and samples. This also applies if ENTIRE VOLUME is specified as the load type.

Loading an individual cue list is done by going to the DISK and selecting CURSOR ITEM ONLY and placing the cursor on the cue list file you wish to load. Pressing F8 - GO - will load the selected cue list into the S3200.

RECORDING AUDIO TO DISK

FEATURES

The S3200 is capable of recording audio to a hard disk. You may record audio whilst sequencing ordinary programs and samples allowing you to effectively overdub onto disk. This may be useful for laying down vocal parts or guitar, saxophone solos and the like over sequenced backing tracks. You may also, of course, playback audio from disk whilst running programs from a sequencer.

The hard disk recording functions include:

1. Full functional operation of the S3200 sampling facilities whilst simultaneously recording to or playing back audio material from a hard disk with no loss of internal RAM.

2. Editing of audio material on disk.

3. MIDI triggering of takes recorded on disk.

4. SONG mode which allows sequential chaining of takes with repeats.

5. Takes may be played alongside programs in a QList.

6. Advanced editing of take parameters that includes level, pan, fade in, fade out.

7. Mono or stereo recording.

8. Varispeed playback of recordings from disk

9. Disk recordings can be processed on an external mixing console by assigning them to individual outputs.

10. Disk recordings can be sent to either of the internal effects units.

11. You can make recordings onto virtually any SCSI hard disk including the Magneto Optical disk.

12. The hard disk can be partitioned to contain a certain amount of space for sound library and a certain amount for disk recordings. The size of each partition may be set by the user. In this way, disk recordings can be associated with programs.

APPLICATIONS

The disk record/playback functions have many applications:

1. Triggering takes whilst sequencing programs. I.e. 'spinning in' backing vocals and the like over sequenced backing tracks either live or in the studio.

2. Music editing in the form of simple 'topping and tailing'
3. Extended remix work using the sequential playback and step repeat capabilities of the SONG functions.

4. In A/V post production when using the Qlist, long dialogue or music tracks can be played from disk whilst simultaneously triggering short, one shot effects and/or looped atmos effects from within programs.

BEFORE YOU GET STARTED - FORMATTING THE HARD DISK

ALLOCATING PARTITIONS

Before you can use the disk record functions, it is necessary to format your hard disk. The S3200 allows you to allocate a certain part of your hard disk for sound library and a certain part for disk recordings. In this way, you may conveniently have takes and sound library on one disk which is particularly useful if you plan to use the disk record functions to 'spin in' recordings over sequenced material because you can have takes and the programs associated with them on one disk.

Pressing DISK and then F6 - FORM - will give you this screen:

```
FORMAT FLOPPY OR HARD DISK : HARD-:
               BLOCKS    HARD PARTITIONS
part.:         good:      size: 60 Mb
 size:          bad:       max: 2

FORMat or ARRange  hard  disk:-> rSTART
LOAD SAVE REN DEL HDSK FORM FORM ARR
```

Using the max: field, you may set the number of partitions you require for sound library storage leaving the rest of the disk free for disk recording.

For example, say you have a 300Mb hard disk - you can allocate maybe 4 x 50Mb partitions for library leaving 100Mb free for disk recording. This would give you 200Mb for library and around 10 minutes of stereo recording at 44.1kHz or twice that in mono. You can, of course, set the disk up as you like depending on whether you want more or less sound library relative to takes.

While formatting, a message shows to say:

```
FORMATTING (typical 10-30 min)........
```

to indicate that the disk is being formatted.

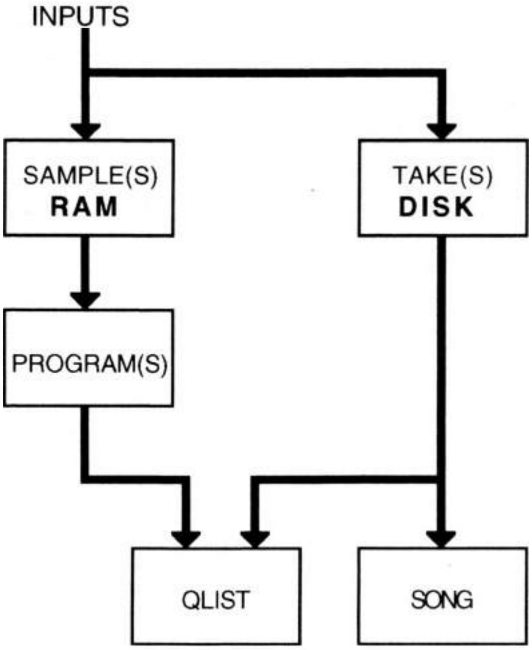
When the formatting process is finished, the screen display will show you how much space is available for disk recording.

> ***VERY IMPORTANT NOTE***
>
> *Formatting the disk will, of course, erase everything on it. MAKE SURE YOU HAVE EITHER COPIED THE CONTENTS ONTO FLOPPY DISK OR HAVE MADE A DAT BACK-UP BEFORE FORMATTING A DISK THAT CONTAINS SOUND LIBRARY. You have been warned!!*

HOW THE DISK RECORD FUNCTIONS WORK

Because the S3200 is equipped with a second LSI (Large Scale Integration) chip to provide the reverb and the second bank of filters, we can use this LSI to provide us with the disk record functions.

The S3200 flowchart is something like this



The takes can be played in a number of ways:

i. Takes can be placed into what we call a SONG and triggered from MIDI. By assigning a take a MIDI note number and channel, incoming MIDI can be used to initiate playback. This can take place whilst ordinary programs are also being played via MIDI and so, in this way, you could, for example, be sequencing several multi-timbral programs as a backing track whilst simultaneously 'spinning in' backing vocals, brass riffs, solos, breakbeats, etc., from disk.

Note: *It is also possible to record a take whilst sequencing programs. For example, you may record a guitar solo directly to disk whilst listening to sequenced programs coming from the S3200.*

ii. Takes can be also be run sequentially in a song and, in this application, several takes can be 'butt edited' to each other and caused to playback sequentially. Furthermore, each step in a song may be set to repeat any number of times so this can be used very effectively for the easy creation of extended remixes.

iii. Takes may also be placed into a Qlist and triggered at SMPTE times. Normal programs may also be placed into the same Qlist. In this way, you could, for example, playback long dialogue or music tracks from disk whilst simultaneously triggering short, one shot sound effects (such as footsteps, etc.) or longer looped atmosphere sound effects from RAM. This will be of particular use to audio visual post production applications.

iv. Takes can be placed into a Qlist for triggering to SMPTE times. The S3200's internal SMPTE generator could then be used to synchronise a sequencer which is playing multi-timbral programs. In this way, the sequencer plays the backing track whilst the takes playback from disk according to SMPTE times. This may be more convenient than using method (i) described above which uses MIDI to play takes alongside sequenced programs.

There are some important things to note when using the disk record functions, however.

1 In order to achieve the disk record functions, the process uses 6 of the the S3200's voices reducing polyphony in this case to 26 voices when recording to or playing back from disk.

2 It is only possible to playback one take at a time from disk - two takes cannot be played simultaneously. If another take is triggered whilst another is playing, the new one will take priority. Crossfades between takes are therefore also not possible.

3 When triggering takes via MIDI, there is always a delay in the take playing back. This is due to disk access time (i.e. the speed with which the disks heads can find the audio material and get it ready to playback). It is possible to accurately set fixed delays so that these can be accommodated when triggering takes from, say, a MIDI sequencer and the sequencers track shifting functions can cater for this.

It is assumed you have a basic working knowledge of the S3200 by now. If you are at all unsure about certain functions, please refer to the appropriate section in this manual for more information.

USING THE DISK RECORD FUNCTIONS

The disk record functions are to be found with the UTILITY pages. Pressing UTILITY gives you this screen:

```
UTILITIES
The following utilities are available:
* ME35T Interface
* Digital Audio
* Cue-lists (smpte)
* Direct-to-disk Recording
DRUM DIGI SMPT  DD
```

F4 - DD - takes you directly to the disk record functions and will display this screen:

```
DD TAKES                  take: TAKE 1
 name: TAKE 1             length: 00:01:00
     *existing take*        type: MONO
 show: MONO                 rate: 44100
total: 00:43:44
 free: 00:44:43            takes:   1
  DD  SONG PLAY EDIT DREC TAKE       DEL
```

Here you may select takes and view their record parameters as well as select new takes for recording or deletion, etc. To return to the main UTILITY functions, press the UTILITY mode key.

The fields on the DD page are:

take: This shows the name of the currently selected take and you may select others by scrolling through them with the DATA control. If this is the first time you have used the disk record functions or you are using a freshly formatted disk, the name field will be blank.

name: This shows the name of the selected take and here you may copy or rename a take. To copy or rename a take, press the NAME key (this field will become highlighted and - new name - will be displayed beneath as soon as a unique name is created) and type in a new name from the front panel and then press ENT. To copy or rename the take, simply press COPY (F6) or REN (F7) as appropriate. If you change your mind, press EXIT to abort the naming process. You may also select takes from here by typing in their names and pressing ENT but remember that the name you type must be the correct one for an existing take otherwise you will be creating a new take. This will be indicated by this field displaying - new name -

show: This allows you to see the free time left on disk or the amount of disk space used expressed as mono or stereo. For example, if you have 10 minutes free on disk when STEREO is selected here, if you select MONO, the free: field (described below) will show 20 minutes.

No other fields are accessible but merely show the takes parameters. These are:

Length:	This shows the length of the currently selected take.
type:	This shows whether the take is a stereo or mono recording.
rate:	This shows the take's sampling rate.
delay:	This shows the MIDI offset delay selected for the recording.

These parameters are explained in detail later in this manual.

The other fields are:

total:	This indicates how much disk space has been allocated for disk recording.
free:	This shows how much space is left on the disk for recording.
takes:	This shows how many takes are on disk. When you use the disk record functions for the first time or use a freshly formatted disk, this field will say 0.

SOFT KEYS IN THE DD PAGE

The soft keys on this page are:

DD	Shows the currently selected page
SONG	This takes you to the SONG mode where you may compile takes for sequential playback or MIDI triggering
PLAY	This takes you to the play pages where you may play takes
EDIT	This takes you to the take editing display
DREC	Takes you to the record setup page.
TAKE	Takes you directly to the record page for recording new takes.
DEL	This allows you to delete a take off disk

At any time, you may play the selected take by holding down the ENT/PLAY key. The take will only play for as long as you hold the key down.

CREATING NEW TAKES

You may create a new take for recording here if you wish simply by typing in a new, unique name. Whether the take is an existing one or a new one will be indicated in the display. It is possible to create new takes in any of the DD function pages simply by typing in a new, unique name.

COPYING TAKES

You may copy takes only within this DD page. This is done by pressing the NAME key, typing in a unique name and pressing ENT/PLAY followed by COPY. The process takes a little longer than actual recording.

RENAMING TAKES

Takes may be renamed only in the DD page. This is done by pressing the NAME key, typing in a new, unique name and pressing ENT/PLAY followed by REN .

DELETING TAKES FROM DISK

You may delete takes from within the DD page. This is done by pressing F8 - DEL. The display will prompt you:

```
! ! DELETE THE TAKE ! ! ??          YES  EXIT
```

Pressing YES will delete the take from disk. Pressing EXIT will cancel the deletion. Be very careful using this feature as deleted takes cannot be retrieved.

MAKING A RECORDING

To prepare for a recording, press DREC in any of the DD pages. You will receive this screen display:

```
DD RECORD SETUP         take: TAKE 1
  mode: MONO          |       free: 00:43:44
source: ANALOG        |     length: 00:01:00
dig.in: ELECTRICAL    |      note: C_3 ch: 16
 start: START SONG    |  stereo: 50 pan: MID
 delay: 400mS         |     indiv: 50  --> 7/8
  DD  SONG PLAY EDIT DREC TAKE BU.L
```

This is the disk record set up page and here you may select and create takes to record as well as set their record parameters. The fields are:

`take:` This shows the currently selected take. To select another, you can either scroll through the takes on disk using the DATA control or type in their names from the front panel by pressing the NAME key, typing in the name and then pressing ENT. You may also create new takes for recording in the same way but by entering a unique take name.

`mode:` This selects whether the recording will be in mono or in stereo.

`source:` This selects the input for the recording - whether it will be through the analogue inputs on the front panel or via the digital audio interface. The selection choices are `ANALOG` or `DIGITAL`.

***NOTE:** If you wish to record through the digital inputs, you will have to adjust the parameters for the digital audio interface as appropriate. These setup parameters for the interface are found in REC1 of EDIT SAMPLE using F8* DIGI.

`dig-in:` This selects whether degital signal is received via the phone jack (`ELECTRICAL`) or `OPTICAL` connection.

`start:` This selects the method by which recording will commence. The options are:

`INPUT LEVEL` - This will cause recording to commence once a certain threshold level has been exceeded. The threshold level is set in the TAKE page (see below).

`FOOTSWITCH1` - This allows you to use the footswitch input to initiate recording.

`MIDI NOTE` - This selects that a MIDI NOTE will initiate recording. The MIDI note number is set in the NOTE field described below.

`M.NOTE+DEL` - This selects that recording will start when it receives a MIDI note but with an offset as set in

the DELAY field described below. The MIDI note number is set in the NOTE field described below.

START SONG - This selects that a MIDI SONG START command will initiate recording.

delay: This allows you to set an offset for the MIDI note reception when M.NOTE+DEL is selected in the start: field.

NOTE: *Because it takes time for a hard disk to actually find the data and play it back, it is necessary to be able to set a fixed offset so that the disk always has enough time to find the take and play it back in sync with any other material that may be playing (for example, when sequencing programs in the S3200). The DELAY field is of use in that you can set the S3200 to start recording after a certain delay when it receives a MIDI note-on. This same note-on can then be used to playback the take in sync with other material.*

By setting a fixed offset of, say, 400mS in the DELAY field and advancing the MIDI note-on in the sequencer by the same amount (i.e. making it 400mS earlier), you can start recording at a predetermined time. After you have made the recording, you can have that take play back from the same point. The SONG mode (described later) always uses these delays to ensure accurate synchronised playback so being able to select to initiate recording with a fixed delay in the RECORD SETUP page allows you to record a take with the offset and then assign it to a SONG (where the offset is always used) without constantly having to re-edit the position of the note on your sequencer.

free: This shows the amount of free time left on disk.

length: Here you may set the length of the recording you wish to make. If you are unsure of the length of the recording you are about to make, simply set a long record time. Wasted disk space can always be edited out and retrieved afterwards in the EDIT page.

note: This field sets the note that will initiate the recording when either MIDI NOTE or M.NOTE+DEL is selected in the start: field. It also sets the note that will trigger playback after it has been recorded. This may be edited after you have made the recording if you wish.

ch: This is an abbreviation of CHANNEL and sets the MIDI channel for the recording when triggering from MIDI. The default is 16 but you may select from 1-16.

stereo: This sets the playback level of the recording. This does not affect the record levels which are set using the front panel REC LEVEL control.

pan: This sets the pan position of a mono recording and the left/right balance of a stereo recording.

indiv: This sets the level of the signal that will be sent to the assignable individual outputs or the internal effects.

--> This sets the destination of the take. The default is OFF but you may set these to any of the individual outputs you want. You will note that the individual output assignment is done in pairs-i.e. 1/2, 3/4, 5/6, 7/8. If the take is stereo then it will be reproduced in stereo through these outputs. If the take is mono, it will appear in mono through both outputs. If, however, the take is mono but you only wish to use one output (perhaps in order to use the other outputs for programs), then select the appropriate pair but set the pan: parameter to L50 or R50 accordingly. If you wish the take to only appear at the individual output(s) you have selected, please set the stereo: parameter to 00 -this will mix the take out of the stereo outputs and so it will only appear at the individual outputs selected here.
You may also assign takes to the internal effects by selecting REB, FX or R+F. In this case, the indiv: parameter sets the effects send level.
You will note that you cannot assign takes to individual outputs AND the internal effects.

At any time, you can audition an existing take by pressing the ENT/PLAY key which will cause it to play back.

SOFT KEYS IN THE DREC PAGE

DD	This takes you to the DD TAKES view page
SONG	This takes you the SONG mode where you may compile takes for sequential playback or MIDI triggering
PLAY	This takes you to the play page where you may play takes
EDIT	This takes you to the take editing display
DD	Shows the currently active page
TAKE	Takes you directly to the record page for recording new takes
BU.L	Takes you to the BACK-UP LOAD page where you may restore takes backed up to DAT.
F8	No function

RECORDING A TAKE

Actual recording is done within the TAKE page. Pressing TAKE - F6 - gives you this screen display:

```
DD RECORD  mode: MONO      TAKE 1
-20dB  free: 00:28:49   length: 00:01:00

 DD  SONG PLAY EDIT DREC METR MOFF ARM
```

This is very much like the REC2 page you will probably already be familiar with in the sample recording pages. It shows you the type of recording you are about to make (i.e. stereo or mono), the take name, the threshold level (if INPUT LEVEL is selected in DREC), the free time left on disk and the length you have set for the new recording. With the exception of the `free:` field, all of these parameters may be changed prior to making a recording. You may also create a new take to be recorded by pressing the NAME key, typing in a suitable name and pressing ENT.

To setup for a recording, set the levels by playing the source to be recorded and adjusting the front panel REC LEVEL control - the incoming signal level will be shown in the bargraph display to the left of the LCD. If you have selected to start recording using INPUT LEVEL in the DREC page, you should set the threshold level by moving the cursor to the field marked `-20dB` and adjusting it accordingly.

To initiate a recording, press ARM. You will receive this screen display:

```
DD RECORD  mode: MONO      TAKE 1
-20dB  free: 00:28:49   length: 00:01:00

WAITING FOR START ......        GO  EXIT
```

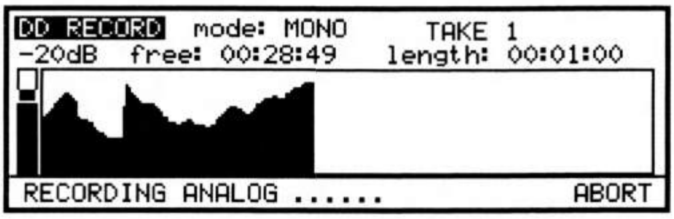
Here, the S3200 is either waiting for a MIDI NOTE or a SONG START command or for the footswitch to be triggered or the input level to exceed the threshold level. This all depends on the type of START you have selected in DREC. You may manually initiate a recording by pressing `GO` (F7). You may cancel this display by pressing `EXIT` (F8).

If the take selected for recording already exists when you press ARM you will receive this prompt:

```
! ! ERASE EXISTING TAKE ! ! ??   YES  EXIT
```

and you may respond accordingly. Pressing `YES` (F7) will cause the existing take to be erased and replaced with the new one you are about to record and pressing `EXIT` (F8) will take you back to the TAKE screen shown above where you may create a new take for recording.

When a recording is being made, the screen shows the incoming waveform as it is being recorded. I.e:



You may stop recording by pressing F8 at any time.

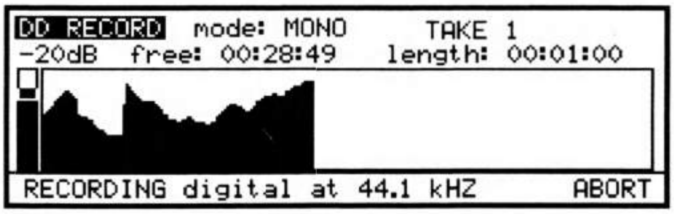
If you are going to record digitally, when you enter the TAKE page, the screen will show:

or whatever sample rate is being received.

If no digital connection has been made or has become disconnected, the display will tell you:

Please check your digital connections.

When recording digitally, the bottom line of the display shows:



and the screen draws the incoming waveform envelope as it is being recorded.

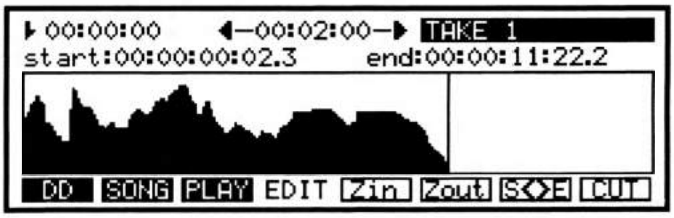
Once a recording has been made, you may use the ENT/PLAY key to play it back.

SOFT KEYS IN THE TAKE PAGE

DD	This takes you to the DD TAKES view page
SONG	This takes you the SONG mode where you may compile takes for sequential playback or MIDI triggering
PLAY	This takes you to the play pages where you may play takes
EDIT	This takes you to the take editing display
DREC	Takes you back to the RECORD SET-UP page
METR	Turns the audio meter on
MOFF	Turns the audio meter off
ARM	This puts the S3200 into a 'record ready' state awaiting the arrival of a suitable record start command depending on the setting of the START field in DREC

EDITING A RECORDING

After you have recorded a take, you may need or want to edit it. This is done by adjusting the take's start and end times and is done within the EDIT page. Pressing F5 - EDIT - in any of the DD pages will display this screen:



When you enter this page, you see a graphic representation of the take's waveform and you may move the start and end points around freely. The start and end points are shown both as time values in the `start:` and `end:` fields respectively and are also shown as flashing vertical lines in the waveform display. You may zoom in or out on a waveform for greater editing accuracy. The fields are as follows:

`▶ 00:00:00`

This shows the 'now' position of the left edge of the screen. By adjusting this parameter, any part of the waveform can be placed as the 'now' position. Used in conjunction with the zoom in and zoom out keys, this function can be used for identifying key points in a take you want to edit without upsetting the start and/or end marks. When the cursor is placed on this field, pressing ENT/PLAY will cause the take to play back from the left of the screen - i.e. the cursor will play from the 'now' position up to the end mark.

`◀—00:02:00—▶`

This shows the 'width' of the screen display in time. Here it is showing that the whole screen width is equivalent to 2 minutes. As you zoom in and out, this field changes to show the equivalent size of the screen.

`TAKE 1`

This shows the name of the take selected for editing. You may select another by scrolling through the available takes.

`start:00:00:00:02.3`

This allows you to adjust the start time of the take. As you adjust this parameter, a flashing vertical cursor moves across the waveform display. You will note that if you move the start point past either extreme of the waveform display, the waveform will scroll. When the cursor is on this field, pressing ENT/PLAY will cause the take to play back from the start point set here.

`end:00:00:11:22.2`

This sets the end point of the recording. As you adjust this field, a vertical flashing cursor moves across the screen. If the end point is moved past either extreme of the waveform display, the waveform will scroll. When the cursor is on this field, pressing ENT/PLAY will cause the take to play UP TO the end point from the point shown on the far left of the screen and you can use the ZOOM IN/OUT functions as a variable pre-roll function. For example, if you want to audition the last

ten seconds or so of a take to check if your end point edit is satisfactory, use the ZOOM keys to set a value close to this in the ◀—00:02:00—▶ field and press the ENT/PLAY key.

You may quickly switch between the start and end fields by pressing the S<>E key.

SOFT KEYS IN THE EDIT PAGE

DD	This takes you to the DD TAKES view page
SONG	This takes you the SONG mode where you may compile takes for sequential playback or MIDI triggering
PLAY	This takes you to the play pages where you may play takes
EDIT	Shows the currently active screen
Zin	Allows you to zoom in on a waveform for greater editing resolution
Zout	Allows you to zoom out of a waveform for a more general overview of the take
S<>E	This toggles the cursor between the start and end marks
CUT	This will cause data either side of the start and end points to be discarded. This is a destructive process so be careful when using this function

PLAYING A TAKE IN THE EDIT PAGE

As in all other DD pages, it is possible to play a take directly by holding down the ENT/PLAY key. In the EDIT page, however, when the take is actually playing, a flashing vertical cursor moves across the screen. When you take your finger off the ENT/PLAY key, playback stops and the play cursor stays at the point where it is stopped. You can use this position to place the start or end mark at a suitable position. There are other differences as discussed above but to recap:

1: If the cursor is on the ▶ 00:00:00 field, pressing ENT/PLAY will cause the take to playback from the point shown on the left of the screen regardless of the setting of the start field. Use this as a means of playing back from anywhere in the take without upsetting edit points.

2: If the cursor is on the end field, pressing the ENT/PLAY key will cause the take to playback from the point shown at the left of the screen up to the end point. You can use this facility to audition the end of an edit and use the ZOOM IN/OUT as a variable preroll function.

USING THE EDITING FUNCTIONS

The editing functions have been kept deliberately simple to use. Basically, you can trim a recordings start and end points and, to assist in this, a graphic representation of the waveform is shown.

Once you have successfully made a recording, the chances are you will want to edit it so go to the EDIT page. Move the cursor to the START field and adjust the start time so that cursor is right at the start of the take. You may use the editing of the start point as well to edit out count-ins to a song or breaths from a backing vocal 'spin in' or whatever. Of course, you can zoom in for greater accuracy.

You may audition your edit at any time simply by pressing the ENT/PLAY key.

Once you have successfully edited the start mark, move the cursor to the END field either by using the cursor keys or by pressing the S<>E soft key and you may set a suitable end point, again, zooming in for greater editing accuracy. Again, use the ENT/PLAY key to hear the result of your edit. Of course, in the case of a very long take, it is very inconvenient to have to hear all of it just to check the end point. When the cursor is on the END field, pressing the ENT/PLAY key will cause the take to play back from the left side of the screen only. In this way, by adjusting the zoom factor accordingly, you may use this as a form of pre-roll so that you only need to audition the last 10 or so seconds of the take in order to assess your end point edit.

HELPFUL HINT: *If you are editing a take for use in a song where takes are sequenced, you might find it helpful to quickly place the edit you are working in an empty song and have it repeat a few times. If it cycles round with no glitches or tempo disruption, then you probably have a good edit that will work well with other takes appended to it. If there is a glitch on the repeat, return to this EDIT screen to fine tune the start and/or end points. The process can be a bit hit-and-miss but as all editing is non-destructive here, it certainly beats using a razor blade!*

Once you have decided that the edit you have done is right, you may wish to discard the unwanted portions in order to free up disk space.

IMPORTANT NOTE: *The discard function is destructive and non-retrievable. Be careful when using it!!*

HELPFUL HINT: *If you are recording in small sections to be compiled as a song later on, please be careful when using the discard function. For example, you may make an edit which, in isolation in the EDIT or PLAY pages, sounds fine but glitches slightly and needs adjustment when playing back-to-back with another take in the SONG mode. If you use the discard function, you may lose the ability to adjust the edit at a later stage. It is recommended, therefore, that you don't use the CUT function until you have successfully compiled your song.*

Of course, you may want to set a start point some way into the recording. An easy way to do this is to press the ENT/PLAY key and let the take play back up to the point where you want to set the start point. When you take your finger off the ENT/PLAY key, the play cursor will stop at that position and you can move the start point to that position. It is unlikely that such an edit will be particularly precise but you can zoom in for more precise editing. The same thing can be done when editing the END point.

PLAYING TAKES

There are a number of ways you can play a take within the S3200's DD functions:

1 You can play a take in ANY page using the ENT/PLAY key

2 You can assign a series of takes to a SONG for sequential playback or for triggering from MIDI

3 You can play 'raw' takes from within the PLAY page and this is accessed in any of the DD pages by pressing F3 - PLAY. Pressing this soft key will give this screen display:

```
DD PLAY/PARAMETERS  take: TAKE 1
samp.rate: 44100Hz  start: M.NOTE+DEL
varispeed: +00.00%  predelay: 400mS
                     note: C_3   ch: 16
  fade in:    10mS  stereo: 50  pan: MID
  fadeout:    50mS   indiv: 50   --> 7/8
  DD  SONG PLAY EDIT DREC TAKE BU.S PRME
```

In the PLAY page, you may audition 'raw' takes using the START option selected during the record process. The fields on this page are as follows:

take: This shows the selected take. You can select other takes for playback by scrolling through them using the DATA control.

samp.rate: This shows the sampling rate at which the selected take was recorded. If the take is an analogue recording, this will always show 44100Hz. If the take was recorded digitally it will show the rate at which it was recorded (i.e. 32kHz, 44.1kHz or 48kHz). You may adjust this in the event that a take recorded digitally was, for some reason, recorded at the wrong sampling rate.

varispeed: This allows you to set the playback rate for the selected take and may be used like a tape machines varispeed control. Normally, this will be 00.00% but, for special effects, you may wish to set a playback rate that is different to the takes sampling rate. This parameter may be adjusted in real-time as the take is playing.

fade in: This allows you to set a fade-in time for the take. The range is 0-9999 milliseconds (i.e. 0 to 10 seconds).

fadeout: This allows you to set a fade-out time for the take. The range is 0-9999 milliseconds (i.e. 0-10 seconds).

start: Here, you can set the method by which a take will commence playback. The options are:

IMMEDIATE - This will cause the take to commence playback as soon as you press PRME - F8.

FOOTSWITCH1 - This will cause the take to start playback when you press the footswitch connected to the footswitch input after PRME is pressed.

MIDI NOTE - This will cause the take to playback when it receives the MIDI note number set in the NOTE field below after PRME is pressed.

M.NOTE+DEL - This will cause the take to play back upon receipt of a suitable MIDI note but with a delay offset set in the delay: field described below. If PRME is pressed, however, playback is immediate in the PLAY page.

START SONG - This will cause the take to playback when it receives a MIDI SONG START command after PRME is pressed.

NOTE: *In all the above selections, the* PRME *key must be pressed before a take is played.*

predelay: This sets the delay offset used for triggering a takes playback when M.NOTE+DEL is selected in the start: field described above.

note: This allows you to set which MIDI note will cause the take to playback when MIDI NOTE or M.NOTE+DEL is selected.

ch: This allows you to set the MIDI channel for the take. In this way, you can set a specific MIDI channel for the MIDI triggering of takes.

stereo: This field allows you to set the playback level of the take.

pan: This allows you to set the pan position of a mono recording or the left/right balance of a stereo recording.

indiv: This sets the level of the signal that will be sent to the assignable individual outputs or the internal effects.

--> This sets the destination of the take. The default is OFF but you may set these to any outputs you want. You may also use this parameter to send takes to the internal effects. The level at which they will appear at the individual outputs or the effects is set in the indiv: field.

SOFT KEYS IN THE PLAY PAGE

DD	This takes you to the DD TAKES view page
SONG	This takes you the SONG mode where you may compile takes for sequential playback or MIDI triggering
PLAY	This takes you to the play pages where you may play takes
EDIT	This takes you to the take editing display
DD	Shows the currently active page
TAKE	Takes you directly to the record page for recording new takes
BU.S	Takes you to the BACK-UP SAVE page where you may archive takes to DAT via the digital audio interface.
PRME	This primes the take for immediate playback from within this page

USING THE PLAY PAGE

Although you may play takes from within any page of the DD functions and whilst the SONG mode is provided either for sequential playback of many takes 'back-to-back' or for setting up lists of takes that you may wish to trigger from MIDI, the PLAY page is provided for playing individual takes in isolation and for setting their playback parameters prior to assigning them to a song.

Assuming you have successfully recorded and edited a take, to play it back from the PLAY page simply press PRME (an abbreviation of 'prime'). This gets the S3200 ready for the incoming MIDI note or song start command by searching for the take selected here. You will receive this screen display:

```
DD PLAY/PARAMETERS            TAKE 1
samp.rate: 44100Hz     start: M.NOTE+DEL
varispeed: +00.00%     predelay: 400mS
                         note: C_3    ch: 16
  fade in:    10mS    stereo: 50   pan: MID
  fadeout:    50mS     indiv: 50    to: 7/8
WAITING FOR START ......           GO  EXIT
```

As soon as it receives the appropriate signal (i.e that set in the START field), it will start playing back. As it is playing, the display will show:

```
playing take:- TAKE 1          ......   STOP
```

You may press F8 - STOP - at any time to stop playback of the take. You may adjust any of the parameters in the PLAY page as you wish and these are automatically saved as soon as you leave this page.

If the take does not play back successfully, the reason is probably that the S3200 has not received the appropriate start command as set in the START field. For example, if MIDI NOTE is selected and C3/Channel 16 is set but, for some reason, the sequencer does not send this note on this channel (i.e. maybe the track set aside for take triggering on your sequencer has been muted or

switched off or the wrong note or channel are being transmitted), the selected take will not play back. Similarly, if you have selected SONG START to trigger the take but your sequencer does not send out a SONG START command in certain modes of operation, the take will note trigger.

> ***NOTE:*** *Akai MPC60 owners will please note that a SONG START command is only issued when the MPC60 is in its SONG mode. If you revert to the main screen to-play an individual sequence, the SONG START selection will not be operative. The same is true of other sequencers so please check your sequencer's manual for more information.*

EDITING A TAKE FOR SYNCHRONISED PLAYBACK

When you make a recording that needs to be synchronised to other audio material (such as sequenced programs, etc.), providing you make the recording referenced to the other audio, there should be no need to worry about synchronisation and the S3200 will follow exactly. There may be instances, however, where some adjustments need to be made and this can be done in the following ways.

1 If the START selection is M. NOTE+DEL, you may adjust the DELAY parameter in millisecond steps to achieve accurate playback start times.

2 If the START selection is M. NOTE+DEL, you may shift the note or track within your sequencer.

3 Whatever the START selection is, you may use the EDIT page to trim a take for better playback start synchronisation.

4 If the take 'wanders' during playback, this will be because the external reference's clock is not stable. For example, if you record a take referenced to an external sequencer playing programs in the S3200 but, after recording has been done, the take wanders out of sync after a few minutes, it is most likely that the sequencers clock is not entirely accurate and so, eventually, the two will drift apart. In this case, use the VARISPEED parameter to adjust the takes playback speed by minute amounts. Unfortunately, there are no guidelines on setting this up. You will have to make an adjustment, see if it improves - if it does not, you will have to try again with a new value. All being well, however, this is only likely to occur if you use a different sequencer to that which was used during the record process.

NOTES ABOUT SYNCHRONISING TO EXTERNAL AUDIO

In most cases, there should be no problems in synchronising takes to external audio material because the playback response time of the S3200 is extremely fast and sync accuracy is in the region of a few milliseconds. Also, the take editing and parameter adjustments described above should overcome any discrepancies you may encounter.

You will note, however, especially when running takes alongside sequenced material that if you change the tempo of the sequence, the takes will be completely out of sync, even if the tempo change is very small. Of course, you can use the VARISPEED control to bring them back in sync but then, of course, they will be out of tune with each other unless you transpose or retune all the sequenced material. Be sure to only make recordings once you have finally settled on the tempo of your sequencer.

Similar considerations must be borne in mind when syncing takes to other audio material such as material off tape. If you use varispeed on the tape machine, you will have to set a suitable varispeed setting on the S3200 to accommodate this. Another consideration is that tape transports are rarely very stable so if you are running a sequencer synchronised to code on tape and also running takes from the S3200's disk, you may find that takes will wander slightly out of sync, especially if that are long recordings.

Hardware limitations in the S3200 prevent it from being able to automatically compensate for changes in sequencer tempo or tape speed.

USING THE SONG MODE

The SONG mode is where you can compile a list of takes for playback. The SONG mode has two functions, in fact. You may use the SONG mode to compile a list of takes for triggering from MIDI notes or you may use the SONG mode to append and playback a sequence of takes 'back-to-back'. The first application, triggering from MIDI, is most likely going to be used when 'spinning in' recordings from a sequencer over other sequenced material. In this way, you may set aside one or more tracks on your sequencer for playing back audio over a sequenced backing track. The other application, sequencing takes, is for creating alternative and extended song remixes.

CREATING A SONG

Regardless of the application you have in mind, the method for creating a song is identical. Pressing F2 - SONG - in any of the disk record pages will give you this screen:

This shows a blank, empty song file. Nothing can actually be done here. To create a song, you must press F2 S.ED to take you to the SONG EDIT page as this is where all the work takes place.

Pressing F2 gives this screen display:

The fields across the top of the screen are as follows:

TL1 — This is the name field for the song. Names of up to twelve characters can be entered here in the normal way by pressing NAME, typing in the name and pressing ENT/PLAY.

nt — NOTE - This sets the MIDI note that the take will be triggered by. This parameter has no function when sequencing takes.

ch — CHANNEL - This sets the takes MIDI channel. This has no function when sequencing takes.

lv — LEVEL - This sets the playback level of the take.

pan	PAN - This sets the pan position of the take if it is a mono recording or sets the left/right balance if it is a stereo recording.
fin	FADE IN - This sets the fade in time for the take and is variable up to 9999mS (10 seconds)
fout	FADE OUT - This sets the fade out time for the take and is also variable up to 10 seconds.
rp	REPEAT - Although not operative when triggering takes from MIDI, this field sets the number of times a take will repeat itself when you run it as a SONG from the RUN key (see below for more information on this function).

You will have to excuse the somewhat cryptic and abbreviated nature of these field descriptions on the S3200's screen but it was felt better to design the screen this way and have all these useful functions available within one page rather than you having to keep switching back and forth between various pages when trying to compile your takes - this would have been highly inconvenient as you can imagine.

NOTE: *Values set in these fields do not affect the 'raw' takes parameters; these always remain the same whatever you set in the SONG mode. In this way, each take in the list can be set to its optimum playback characteristics in a song without affecting the raw take.*

Soft keys F1 to F4 take you to different pages. The soft keys F5 to F8 have the following functions:

MARK	This marks a step in the song for inserting or deleting.
BLCK	This allows you to mark a block of cues for inserting or deleting.
INS	Pressing this will insert the marked step or block at the point of the cursor on the scroli bar.
DEL	Pressing this will delete the marked step or block.

To create a song, whether it is for MIDI triggering or for sequential playback, the method is exactly the same. Follow these steps to create a song.

In the SONG EDIT page, move the cursor to the first empty take field. This is done by moving the cursor one position to the right. You will have a screen something like this:

```
 1 TL1              nt ch lv pan fin fout rp
                    C_3  1 50 MID   10   50  1



  DD  SONG PLAY EDIT MARK BLCK INS  DEL
```

You may now select a take using the data control (or the +/< or -/> keys found on the numeric keypad). Having done that, the selected take's

parameters will be loaded into the step and so the note, level, pan and other fields may change if the raw takes parameters are different from those set as the default shown above. You may change these as necessary. Once you have assigned your first take, you should receive a screen something like this:

```
  1 TL1            nt ch lv pan fin fout rp
   B.VOX 1         C_1 16 65 R40   20  50  1



  DD  SONG PLAY EDIT MARK BLCK INS  DEL
```

If you are triggering from MIDI, you will most likely want to edit the MIDI note and channel and if you are creating a song to run takes sequentially, you will possibly want to set a repeat for that step. If you are at all unsure of the take you have selected, you can press the ENT/PLAY key to audition the selected take.

There are several ways you can create the next step but the easiest method is probably this:

Press F5 - MARK. This will mark the first step and a small block will appear beside it. Now press F6 - INS(ert) - and this will copy that step and you will see it appear beneath the first step. Now press 1 on the numeric keypad to move the pointer down to the new step. You may now move the cursor into the take field and select another take as the next step in the list. Of course, you may edit that take's parameters if you wish. Repeat the above:

Press MARK - press INS - press 1 - move the cursor to the take field to select the next step - edit the parameters accordingly using ENT/PLAY to audition.

NOTE: *If the cursor is not on the scroll bar down the left hand side of the screen, pressing ENT/PLAY will first return the cursor to the scroll bar. Pressing it again will then play the selected step. Do not be alarmed if pressing the ENT/PLAY key the first time does not play the take.*

You may repeat that process as many times as you like until you have created your song.

NOTE: *If you are creating a song for triggering takes from MIDI, the order in which the takes appear is not important. It is probably just as well to keep some semblance of order, however, so that it is easier for you to keep track of. For example, it seems pointless putting the last chorus's backing vocal in first and the first verse last!*

If, however, you are creating a song for sequential playback, it is necessary to assemble the takes in the order you wish them to play back.

If, at some point, you wish to delete a take from the list, simply move the cursor to the required step, press MARK and press DEL.

NOTE: *If you do not press MARK you may find that you delete the wrong step. Be careful because although this is not ultimately destructive (after all, it's easy enough to insert the step back in again) it can be annoying.*

You may give the song a name (if you haven't already) and then save it to disk.

It really is quite simple and whatever your application, whether it be sequencing takes or triggering them, you will soon build up quite complex lists very quickly. Advanced editing such as block editing, copying, deleting and shifting are explained later. You will also want to refer to the section USING THE KEYPAD TO GET AROUND THE SONG which appears later in this section. For the time being, practice the above until it becomes a natural process.

TRIGGERING TAKES FROM MIDI

This powerful function of the S3200's disk recording capabilities allows you to simultaneously playback audio from the hard disk whilst sequencing programs in the S3200. To set a up a list of takes for MIDI triggering, press the SONG key (F2 in all disk record pages) to get the following screen:

```
1 TL1                nt  ch lv pan fin fout rp
                   ? C_3  1 50 MID   10   50  1




 DD  S.ED PLAY EDIT DREC TAKE        RUN
```

This shows a blank, empty song or take list. To create a new list of takes, press F2 again. This takes you to the S.ED or SONG EDIT page where you may assemble your list. Pressing S.ED gives you this display:

```
1 TL1                nt  ch lv pan fin fout rp
                   ? C_3  1 50 MID   10   50  1




 DD  SONG PLAY EDIT MARK BLCK INS  DEL
```

Here we can see that the first take is blank and has some default parameters assigned to it. Assign your takes as described above and edit the parameters if necessary.

USING MIDI TRIGGERING

The method for assembling a list of takes for triggering from MIDI is described above. Once you have set up a series of takes for MIDI triggering, simply sending the appropriate MIDI notes on the selected channel(s) will cause them to play back. Typically, a list of takes for MIDI triggering may look something like this:

```
1 TL1                nt  ch lv pan fin fout rp
  B.VOX 1            C_1 16 65 R40   20   50  1
  B.VOX 2            C#1 16 55 L10    0    0  1
  GUITAR SOLO        C_3 16 60 MID   10   12  1
  BRASS RIFF1        C_4 16 78 MID    0   20  1

 DD  S.ED PLAY EDIT DREC TAKE        RUN
```

Here we can see a typical setup for spinning in material over a sequenced backing track. We can see that there are two backing vocal recordings which will trigger when they receive C1 and C#1 and a guitar solo will start to playback on C3 with a brass riff playing off C4. All the takes are on MIDI

channel 16 although, in practice, there is nothing to stop you setting different MIDI channels for some or all of the takes. For example, you could have set the backing vocals to MIDI channel 15, the guitar solo to MIDI channel 14 and the brass riff on MIDI channel 16. In this way, you can reserve tracks on your sequencer especially for certain audio parts. This may be useful if you need to slip parts using the track shift function on your sequencer.

In the above example, you can see that some takes have fades set for them. This facility is useful for 'softening' the start and end points of a take whose edit may be a bit abrupt. The range for both fade in and fade out times are 10 seconds (actually 9999 milliseconds but who's counting one millisecond!). To soften an abrupt attack or end, fades of around 5-20 milliseconds will normally do the trick. Fades longer than that can be useful for fading in a take or causing a smooth, gradual decay at the end of a take's replay.

As soon as the S3200 receives the appropriate note on the appropriate channel (i.e. one that is assigned to a take in a song, the following screen will be displayed:

```
1 TL1           nt  ch lv pan  fin fout rp
  B.VOX 1       C_1 16 65 R40   20   50  1
  B.VOX 2       C#1 16 55 L10    0    0  1
  GUITAR SOLO   C_3 16 60 MID   10   12  1
  BRASS RIFF1   C_4 16 78 MID    0   20  1

playing take:- B.VOX 1        ......  STOP
```

You may either issue a MIDI ALL NOTES OFF command from your sequencer/keyboard or specifically press F8 - STOP - on the S3200.

NOTE: *To play takes from MIDI in this way, the takes MUST be set to start from MIDI NOTE or M.NOTE+DEL in the PLAY page.*

When playing back in this way, triggering from MIDI, there is ALWAYS a fixed delay and the delay is that set in the DREC or PLAY pages. Normally, it is probably best to offset the delay by the same amount for each take. By doing this, you can shift one or all tracks on your sequencer by a consistent amount. If you wish, however, each take may be set to have its own unique offset delay which may help in syncing up some takes. Remember that you can use a combination of your sequencer's track shifting and the variable delay parameter to get takes exactly in sync. For example, your sequencer's track shifting functions may not offer enough resolution to obtain precise triggering of the take(s). In this case, adjust the take's delay time in milliseconds to obtain precise sync.

HELPFUL HINT: *If your sequencer does not have a wide enough range for shifting a track, why not insert a blank bar or half bar at the beginning of the sequence and then delete that bar only on the track(s) devoted to triggering takes. You can then use the S3200's MIDI delay to offset the triggering time.*

One thing to remember, of course, is that if you wish to trigger the same take several times (i.e. in the case of a backing vocal you wish to spin in over every chorus), you do not have to specify it in the list several times when triggering it from MIDI. You only need to select it once and, when the S3200 receives the appropriate note, that take will play.

You will note that, when triggering takes from MIDI, the repeat field has no function - this is for use when running takes sequentially in a song (see below).

IMPORTANT NOTES ABOUT TRIGGERING TAKES FROM MIDI

When triggering takes from MIDI, if you stop the sequencer and restart it somewhere in the middle of where a take should be playing, it will not sound. This is because it requires the MIDI note-on to trigger it. In such circumstances, it will be necessary to 'rewind' the sequencer to a point somewhere before the MIDI note to ensure that the take receives the note on and will trigger. The same, of course, is true if you are running your sequencer synchronised to tape and triggering takes from MIDI. If you stop the tape, you will need to rewind it to a point before the MIDI note required to trigger the take

When a take is triggered from a MIDI note, when you stop the sequencer, assuming your sequencer sends out a MIDI ALL NOTES OFF command, the take will stop playback. If however, your sequencer does not issue an ALL NOTES OFF, the take will continue to play but you may specifically stop playback from the S3200 front panel using F8 - STOP.

Because the S3200 can only play one take at a time, crossfading between takes is not possible. Also, if one take is playing while another is triggered, the new take will take priority although please note that there will be a short gap between the one take finishing and the next take starting.

USING THE SONG MODE TO CHAIN TAKES

This mode of operation allows you to playback takes sequentially 'back-to-back'. This mode will be invaluable to remix engineers for the creation of extended remixes. It will also be of use to jingle writers and TV theme music writers who often have to provide several versions of one piece of music with different lengths.

The simplest method of using the song mode is to 'top and tail' an entire recording (i.e. record an entire song into the S3200 and edit its start and end points) and have it play back from within the SONG mode. At a more advanced level, you could use the SONG mode to play back a whole series of entire songs recorded and edited this way and the SONG mode can be a convenient method of sequencing the tracks on an album. A more advanced application for the song mode, however, is to create extended remixes.

A typical song will look something like this:

1 TL1	nt	ch	lv	pan	fin	fout	rp
INTRO 1	C_3	16	65	R40	20	50	1
INTRO 2	C_3	16	55	L10	0	0	2
INTRO 1	C_3	16	60	MID	10	12	1
VERSE 1	C_3	16	78	MID	0	20	1
BREAK	C_3	16	60	MID	10	12	4
DD	S.ED	PLAY	EDIT	DREC	TAKE		RUN

Normally, the takes will be recorded onto disk in sections, one by one, edited and then compiled into a song within the S.ED page. Here we see a series of such takes running 'back-to-back' to form a typical extended remix. You can see that some steps are set to repeat several times and the repeat field can be

used to good effect in this way - it certainly beats having to print several versions of a section onto tape and splicing them all together!!

Crossfades are not possible in the song mode but this isn't such a big disadvantage for most remix work where the material is usually quite percussive and butt editing can be very effective. Of course, a good edit depends on the accuracy of the cuts you make in the EDIT page and hearing a cut out of context may not always give you a true impression of how it will sound back to back with another cut. A typical editing session will probably involve a bit of switching between the EDIT page and the SONG page to fine tune some edits which, on their own sound fine but alongside other takes, exhibit some form of glitch. When you switch from the SONG page to the EDIT page, the take just played will be in the EDIT window and then, when you return to the SONG page, the step you are working on will still be current so a fine edit should only take a few seconds. Often, the fade in and fade out parameters can come in useful for smoothing out such problems. Experimentation is the name of the game here!

Once you have a few takes in your song, pressing RUN (F8) in the main SONG page will cause them to play back sequentially. When this happens, you will receive the following screen display:

```
 1 TL1           nt  ch lv pan  fin fout rp
  INTRO 1        C_3 16 65 R40   20   50  1
  INTRO 2        C_3 16 55 L10    0    0  2
  INTRO 1        C_3 16 60 MID   10   12  1
  VERSE 1        C_3 16 78 MID    0   20  1
  BREAK          C_3 16 60 MID   10   12  4
PLAYING TAKE-LIST ........       STOP SKIP
```

Pressing F7 will stop playback of the song and pressing F8 - SKIP - will cause the song to skip the current step and proceed playback from the next step.

NOTE: *The SKIP function causes playback to start from the next step, not the next repeat of any step.*

As the song is playing, a small highlighted box appears to the left of the step to indicate your playback position. If the song exceeds five steps, the list of steps will scroll up the screen with the currently playing step being placed in the centre of the screen. The step number at the top left of the screen also changes to show the step currently playing. If a series repeats have been set for a step, they will count down as they are played so you can easily keep track of progress during playback.

You may play from any position in the song simply by moving the pointer up or down the scroll bar. This can be done using the DATA control or the numeric keypad (see later USING THE NUMERIC KEYPAD). Pressing RUN will cause the song to commence playback from that step. At the top of the screen to the left of the song name is an indicator showing which step you are on.

ADVANCED EDITING

USING THE NUMERIC KEYPAD IN THE SONG MODE

The numeric keypad can be used as a shortcut method of moving around the song steps. You can use the cursor keys if you wish, especially if you are only moving one step down but when you wish to move several steps down or up, the keypad will become very useful to you.

The shortcuts are simple and easy to remember:

With the cursor on the scroll bar...

pressing 1 on the numeric keypad will move you one step down

pressing 2 will move you two steps down

pressing 3 will move you three steps down

pressing 4 will move you four steps down and so on through to 9 which, of course, will move you nine steps down.

pressing the -/> key and 1 simultaneously will move you UP one step

pressing -/> and 2 will move you up two steps

pressing -/> and 3 will move you up three steps

pressing -/> and 4 will move you up four steps and so forth through to -/> and 9 which will move you up 9 steps.

Other keys you can use are:

0	This will take you to the first step in a song
+/<	This will take you to the last step in a song
ENT/PLAY	This will always move the cursor to the scroll bar

Please note, however, that, with the exception of the ENT/PLAY key, these keypad shortcuts ONLY WORK WHEN THE CURSOR IS ON THE SCROLL BAR. Using them when the cursor is on a parameter field in the S.ED page will cause them to input a numeric value.

BLOCK EDITING IN SONG MODE

So far we have seen how to create songs in a fairly simple fashion - i.e. in the S.ED page, press MARK, press INS, move the cursor down a step, select a new take - and this will get you through creating a song. Even if that is all you learn to do, you should find song creation quite straightforward. There are other editing techniques available in the SONG mode, however, that makes this mode even more powerful.

Using the MARK and BLCK keys, you may identify whole blocks of steps and copy and shift them around a song very conveniently. The easiest way to explain this is by example. Let us say you have the following:

TAKE 1	This is a 1 bar drum section with a cymbal at the down beat of bar 1

TAKE 2	This is a 2 bar drum section without the cymbal
TAKE 3	This is a 1 bar drum beat with a small fill at the end

You have set the steps up as follows:

TAKE 1	No repeat
TAKE 2	3 repeats
TAKE 3	No repeat

You have created a 8 bar drum section complete with a cymbal at the start and ending. Let's say you now want that whole section to repeat 4 times. You could do it the hard way and mark and insert each step sequentially but the easiest method is this:

Ensuring the cursor is on the scroll bar, move the cursor to the first step in the block, TAKE 1, and press MARK. Now move the cursor down two steps by pressing 2 on the numeric keypad and press BLCK. This marks the three steps as one block. Now move the cursor down a step (press 1 on the keypad) and press INS. You will copy that block at the end of TAKE 3.

You can now do one of two things to copy it twice more. You can either move the cursor to the end of the second block (press 3 on the numeric keypad) and press INS again and then move it to the end of the third block (press 3 again on the keypad) and press INS once again. The other way to achieve the same effect is to move the cursor back to TAKE 1 and press MARK and then move the cursor to the second occurrence of TAKE 3 (press 5 on the numeric keypad) and press BLCK, press 1 on the keypad to move the cursor down a step and press INS. This will append the whole block onto the end of itself. Either way is equally effective so choose whichever is easiest for you.

This method of block copying and inserting has even more uses than just appending block onto themselves. Again, another example should demonstrate this.

Let us say you have the three takes sequenced as above - TAKE 1 once, TAKE 2 three times, TAKE 3 once - and you wish to insert this just before the BREAK in your extended mix. Move the cursor to TAKE 1 and press MARK. Press 2 on the numeric keypad to take you to TAKE 3 and press BLCK. Now move the cursor to the point just before BREAK - i.e. with the arrow pointing just above it thus:

```
1 TL1           nt  ch lv pan fin fout rp
  TAKE 1        C_3 16 65 R40  20   50  1
  TAKE 2        C_3 16 55 L10   0    0  3
  TAKE 3        C_3 16 60 MID  10   12  1
  VERSE 1       C_3 16 78 MID   0   20  1
  BREAK         C_3 16 60 MID  10   12  4
 DD  SONG PLAY EDIT MARK BLCK INS DEL
```

Now press INS and the whole block will be inserted before the BREAK.

You will find these editing functions extremely useful when creating complex remixes as they allow you to shift whole sections and place them anywhere you like in a song very quickly and easily. It means that you can regard what are small, short takes appended together as one long take which can be inserted as you wish.

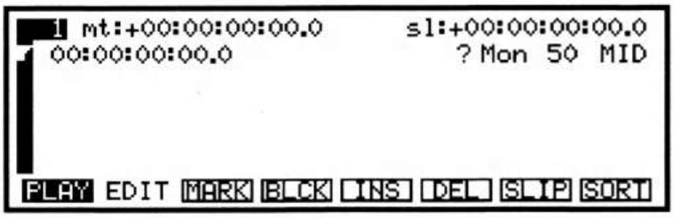
> ***NOTE:*** *It is only possible to mark contiguous steps as a block - you cannot mark a series of steps, skip a few and then mark another few steps as one block.*

Naturally, this block editing function can be used to delete blocks of steps as well.

USING TAKES IN QLISTS

Takes can also be played from a Qlist. To assign a take to a step in the Qlist, go to the UTILITY mode and press F3 to take you to the QLIST pages. Assuming you have not already loaded in an existing Qlist from disk, you will receive a blank Qlist display thus:

Now press F2 - EDIT - to display this screen:



By moving the cursor to the field where it shows `Mon` , you may select from the following options:

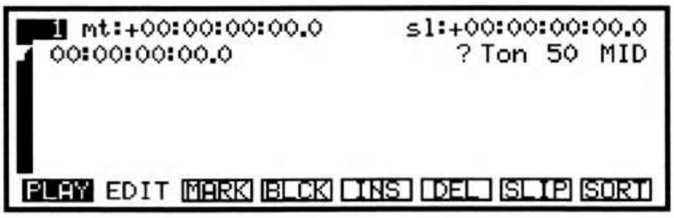
`Mon` — This will cause a program to be triggered at the note and velocity value at the shown SMPTE time for that step.

`Mof` — allows a program that may be sounding (i.e. a looped sample) to be turned off (i.e. stop sounding) at the shown SMPTE time for that step.

`Ton` — allows a take to be triggered at the shown SMPTE time for that step.

`Tof` — allows a take that may currently be sounding to be turned off at the shown SMPTE time for that step.

As soon as you change `Mon` to `Ton` , the display changes:



The note number and MIDI velocity parameters are replaced with level and pan parameters for the take. In this way, you can conveniently set individual level and pan for every cue in the list.

For more information regarding setting up, creating, editing and playing Qlists, please refer to the section in this manual that explains the Qlist functions. All other functions remain the same with the exception of the above

which has been introduced to differentiate between programs to be triggered and takes to be played back from disk.

USING THE QLIST

If you have already used the Qlist facilities in the S3200, you will already know how to create, edit and play back Qlists. You will have used this to trigger samples in programs relative to a specific timecode position. You may also trigger takes directly from disk.

To do this, in QLIST EDIT, simply insert a cue into the list and switch `Mon` to `Ton`. When you move the cursor to the name field, you will now only be able to select takes for that cue. Set a suitable time for the cue and, in QLIST PLAY, press Pint or Pext and the take will play once the appropriate SMPTE time is reached. You may continue this process until all your takes and programs are assigned and you may freely mix programs and takes to create a Qlist something like that shown in the screen below:

```
QPLAY  QL1                 time:    :  :  :  .
 00:00:04:21.5 DOOR SLAMS1     Mon C_3 127
 00:00:12:12.4 FOOTSTEPS       Mon C#3 127
 00:00:13:10.6 BG MUSIC 1      Ton  50 MID
 00:00:14:05.1 FOOTSTEPS       Mon C#3 127
 00:00:14.16.7 GUNSHOT 1       Mon C_3 127
PLAY EDIT SMPT GRAB Pext Pint Cint STOP
```

Here we have some sound effects being played from within programs and at around 13 seconds, some background music is triggered from disk. Of course, at any time, you may edit a SMPTE start time to ensure precise synchronisation of takes to SMPTE.

This method of operation is probably of most use to A/V engineers for placing long music tracks or dialogue plus sound effects to picture - in this case, the sound effects are being triggered from RAM via programs and the music and/or dialogue from the hard disk.

NOTES ON USING THE QLIST

Not only can takes be triggered to start playback at specific SMPTE time within the Qlist, they can also be made to stop at specific SMPTE times. To do this, simply insert a cue and set that to `Tof` and select the appropriate take. For example:

```
 1 mt:+00:00:00:00.0      sl:+00:00:00:00.0
 00:00:04:21.5 DOOR SLAMS1     Mon C_3 127
 00:00:12:12.4 FOOTSTEPS       Mon C#3 127
 00:00:13:10.6 BG MUSIC 1      Ton  50 MID
 00:00:14:05.1 BG MUSIC 1      Tof  50 MID
 00:00:14.16.7 GUNSHOT 1       Mon C_3 127
PLAY EDIT MARK BLCK INS DEL SLIP SORT
```

Here, the same take is specified twice but one is specified as tof which indicates that at 00:00:14:05.1, the take will stop playing. In this way you can set exactly how long a take is to play for regardless of any edits made to that take within the DD pages of EDIT SAMPLE.

You may, at any time, change a take cue into a program cue simply by changing `Ton` to `Mon` and vice versa.

If you try to start playback of a Qlist midway through where a take should be playing, you will note that the take will pick up from that point regardless of where it is started. You will note, however, that this will not happen with MIDI cues (i.e. triggered programs) which need to wait until reception of the next MIDI note on command.

If two takes overlap in a Qlist, when the second (i.e. the later cue) plays, the first will be cut giving priority to the latest one triggered.

NOTE: *When overlapping cues are played in this way, there will be a small gap between the end of the first and the start of the second cue. Similarly, if two cues are set to butt together, you will notice a small gap between them.*

For more information on the use of the Qlist, please refer to the appropriate section in the S3200 Operators Manual.

LOADING AND SAVING SONGS AND QLISTS TO DISK

Only one song or Qlist can reside in the S3200's memory at any one time but you may save as many songs and Qlists to disk as you like. This is done in the normal way in the DISK mode just as though you are saving or loading individual programs, samples or effects files.

Press the DISK key and you will receive the usual disk page. To save a song, select CURSOR ITEM ONLY and move the cursor to the song name. If you have given your song a name this will be displayed and you may save it to disk. If you have not given your song a name, the default TL1 will be displayed and this may be saved in the usual way. Songs may be loaded by selecting LOAD, moving the cursor to the song and pressing GO. The same, of course, is true of Qlists and these may be saved and loaded in the same way. Of course, you may wish to load or save an entire volume in which case the song and/or Qlist will be loaded or saved along with any other material in the S3200's RAM.

NOTE: *Any number of songs or Qlists may be saved to disk but they must all have unique names. If you create one song or Qlist and then create an alternative version, be very careful to give it a new, unique name otherwise, should you try to save the new version without giving it a new name, the other version will be overwritten.*

ARCHIVING DISK RECORDINGS

BACKING UP TAKES TO DAT

Back up of takes to DAT is done in the PLAY page of the disk record functions. The soft key F7 - BU.S - takes you to the BACKUP SAVE page. I.e:

```
DD PLAY/PARAMETERS  take: TAKE 1
samp.rate: 44100Hz   start: M.NOTE+DEL
varispeed: +00.00%   predelay: 400mS
                       note: C_3   ch: 16
  fade in:   10mS   stereo: 50   pan: MID
  fadeout:   50mS    indiv: 50    to: 7/8
  DD  SONG PLAY EDIT DREC TAKE BU.S PRME
```

Press F7 - BU.S - will display this screen:

```
TAKE BACKUP SAVE     take: TAKE 1

  transmit rate: 44.1kHZ

                        ,save,
  DD  SONG PLAY EDIT ONE  ALL  BU.S STOP
```

This simple page displays the transmit rate for the back up to DAT. This can only be set to 44.1kHz and so this field is not accessible to the user but is shown merely for information. When a take is backed up to disk, all data related to it is also backed up, of course. This includes start and end edit points, MIDI parameters, fade in/out, etc..

To back up takes to DAT, you can either do them individually or all of them. To back up a single take, select the appropriate take in the take field at the top right of the screen. Press RECORD on your DAT machine and then press F5 - ONE . This will back up only the selected take to DAT.

To back up all the takes on disk to DAT, irrespective of the take selected in the take field, press RECORD on the DAT machine and then press F6- ALL. The S3200 will then systematically back up all the takes contained on that disk. When either of the above is taking place (i.e. a single take or all takes are being backed up), you will receive the following display:

```
 saving take:- TAKE 1          ......  STOP
```

This shows the current take being backed up. At any time, you may abort the procedure by pressing F8 - STOP.

RESTORING TAKES BACK FROM DAT TO DISK

The take restore functions are found in the DREC page on F7 - BU.L. I.e:

```
DD RECORD SETUP      take: TAKE 1
  mode: MONO        |        free: 00:43:44
source: ANALOG      |      length: 00:01:00
dig.in: ELECTRICAL  |      note: C_3 ch: 16
 start: START SONG  |  stereo: 50 pan: MID
 delay: 400mS       |   indiv: 50   --> 7/8
  DD  SONG PLAY EDIT DREC TAKE BU.L
```

Pressing this gives this screen display:

This page looks very much like the normal record page except that you have lost the threshold field. To perform a restore of either a single take or all takes, line the DAT up to the appropriate point and press F8 - ARM on the S3200. You will receive the following message:

```
name-match takes will be wiped! OK   EXIT
```

indicating that any takes currently on disk that have the same name as those backed up on DAT will be overwritten and replaced with those from DAT. Press OK or EXIT as appropriate.

IMPORTANT NOTE:

PRESSING 'OK' WILL IRREVOCABLY ERASE THOSE TAKES THAT HAVE THE SAME NAME.

BE VERY CAREFUL IN YOUR SELECTION

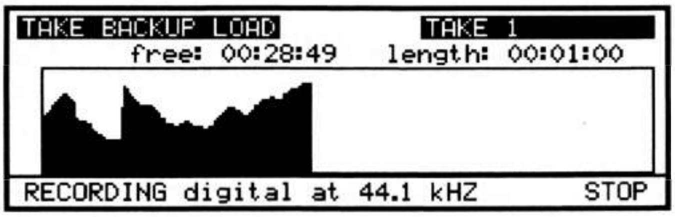
Immediately after you press OK, the display will show:

```
carrier - 44.1kHZ
waiting for take from DAT....        ABORT
```

This indicates that the S3200 is receiving the digital signal through the digital interface. You should now start your DAT machine.

If the display shows WAITING FOR CARRIER, this indicates that the correct digital connection has not been made. Please check your connections and also please check the settings of the digital interface in the DIGI page of REC1 in EDIT SAMPLE. You may press F8 - ABORT - to exit this screen.

Assuming you wish to proceed with the restore, press PLAY on your DAT machine. When the take is being restored to disk, the screen will show:



This is virtually identical to the normal record page when a recording is being made and the takes waveform envelope will be displayed as it is being restored. The DAT restore function is, actually, just another way of recording and this 'confidence monitoring' keeps you informed of progress during the

DAT restore process. At any time, you may press F-8 STOP to abort the restore.

APPLICATIONS FOR DISK RECORDING

So far we have seen the functions of the disk record and playback facilities offered by the S3200. Let us now look at some typical applications and how these functions might be used.

REMIXING

The basic premise of the disk functions for remixing is very simple. Make your recordings in sections, trim the start and end for each section and then chain them together in any order you wish with or without repeats for each step using the sequential SONG mode. For example:

Let us say you are mixing down a track for an extended remix. Typically, you will mix each section separately, muting some channels on your mixer, opening up other channels and so forth. For example, you may be working on the intro and have only the basic instruments of drums, bass and some guitar. Give the take a new name and record this onto the hard disk. Now, rewinding the master multi-track to the start of the intro again, open up some channels so that, for example, some keyboard parts are introduced. Give the take a new name and record that onto the hard disk. You may then want to go straight to an accappella chorus so locate the master multi-track to that position, mute all channels except the vocal parts you want, name the new take and record that onto the hard disk. At this point, you may want the intro to come back in so rewind to that and set the channels as necessary and record it onto hard disk or, of course, simply use an existing version of the intro you have already recorded in the SONG mode later on. Keep on doing this until you have all the sections separately recorded on disk. At any time, you may audition the recording you have just made using the ENT/PLAY key.

You should now edit the start and end points in the EDIT page using the functions described above. You may audition your edits using the ENT/PLAY key. Having done that to your satisfaction, go to SONG, press F2 - S.ED - to take you to the SONG EDIT page and assign the takes in the order you want them to play back with or without repeats as you wish. You may go directly to the main SONG page simply by pressing F2 again to hear the results of your efforts. If some edits are not smooth, move the cursor to the offending step and then go directly to the EDIT page where you may trim the start and/or end points again (you will probably need to zoom in quite a bit to set these times accurately) and return to the SONG page. The cursor will still be on that step so you can immediately check it using the RUN key (F8).

Of course, you don't have to record the sections in the order you want - you may record them in any order and then sequence them later within the SONG mode.

Having done all this, save the SONG to disk. You may only have one song in the S3200's memory at one time but you may save as many as you wish to one disk. In this way, you may have several alternative remixes saved to disk which can be loaded very quickly for comparison purposes.

HELPFUL HINTS WHEN REMIXING

1. Record a second or so of material either side of the section so that you have 'handles' either side of the recording with which to play with when editing.

2. Don't use the CUT function to discard unwanted audio either side of the edit until your SONG is fully assembled. Although the edit may sound good in isolation in the EDIT page, you may need a bit of leeway to play with when sequencing the takes in the SONG mode. Using the CUT function too early may prevent this and so you may need to re-record that section again if you cannot get a smooth transition between steps.

3. To check if an edit works or not, quickly move to the SONG mode and assign the take you are editing and have it repeat several times. If the take loops round smoothly then you have created a good edit which should also work well sequenced back-to-back with other steps. It is a good idea to do this for every edit to ensure that edits are smooth. This 'sketch pad' song can be replaced or erased later.

TRIGGERING TAKES FROM MIDI

Typically, this method of playback of disk recordings will be used to spin in such things as backing vocals, solos, etc., over sequenced material and, of course, as you know, you may simultaneously be sequencing programs in the S3200 whilst recording to disk or playing back from disk.

It is, of course, possible to actually record to disk whilst simultaneously sequencing programs in the S3200's memory and this powerful feature allows you to drop into record over sequenced material while it is playing. This can be used to record a guitar part or vocal line over sequenced material. To do this, ensuring that your programs are all properly selected for playback from your sequencer in the SELECT PROG display, go to the DD functions and select DREC. Set up the recording as appropriate.

As an example, let us say you wish to record a short guitar part over part of your sequence. Locate the sequence to the required point. Now there are several ways in which you can set the recording. You may wish to set it so that the sequencer triggers recording by sending out a MIDI note at a suitable point - you may like to use the M. NOTE + DEL function as this can then subsequently be used to trigger playback later on without you having to edit your sequencer. You could select INPUT LEVEL, however, and simply drop into record manually or you could use the footswitch selection for hands free operation.

So, locate the sequencer to the appropriate point (give yourself a few bars intro to get ready) and press PLAY on the sequencer. Depending on the type of START selection you have made, make your recording playing along with the sequenced material. When you have finished, stop the recording.

At this point, you may wish to edit the start and/or end point to ensure a clean edit. Having done that, go to the SONG mode, press F2 to get to S.ED and assign the take. You may select the MIDI note number and channel here as well as set fade in and out times and level and pan position.

NOTE: When you assign the take to the SONG for playing back from a MIDI trigger, to ensure an accurate and consistent trigger, you must select M.NOTE+DEL. If you do not, the take may trigger erratically and slightly at the wrong time. This is because of disk access speeds.

You must, of course, also offset the MIDI note in the sequencer used to trigger this recording. Using a combination of your sequencers track shift and the S3200's predelay function, you should be able to get very accurate and precise triggering.

HELPFUL HINT: You can set up the song BEFORE you actually make the recording. For example, let us say you wish to overdub a short guitar solo over the top of sequenced material. Go to the S.ED mode and move the cursor to the take name field where you will want the new recording to be assigned after it has been made. Press NAME and type in a suitable name into that field - i.e. GUITAR 1. A question mark will be shown to indicate that the take does not yet exist. Now go to the DREC/TAKE pages and name the recording you are about to make exactly the same (i.e. GUITAR 1). Now make the recording. Once the recording is completed, you may go straight to the SONG page and have it playback immediately.

PROCESSING DISK RECORDINGS ON AN EXTERNAL MIXER

There are times when it might be desirable to process disk recordings externally. For example, on a vocal line triggered from MIDI you may wish to add some reverb or EQ or whatever. The disk recordings may be assigned to any pair of individual outputs - 1/2, 3/4, 5/6, 7/8 - and so these can be routed through an external mixer for processing. If the recording is mono, then it will only appear at the odd numbered output (i.e. 1, 3, 5 or 7). To stop them also appearing at the left/right mix outputs, turn the stereo: field to 00 in the PLAY pages. You will need to specifically set this for every take you wish to process.

Of course, you may prefer to use the S3200's internal effects in which case, the disk recordings may be assigned to RVB (to send it to the internal reverb) or FX (to send it to the echo, chorus, delay or pitch shift effects). You may also select R+F to send it to both sections simultaneously.

EDITING DAT RECORDINGS

Although they offer superb sound quality and are relatively inexpensive these days, the big problem with DAT machines is that it is impossible to edit them. When mixing down directly onto DAT this can be a real problem. You can use the S3200's disk record function to edit DAT recordings simply by transferring the recording into the S3200, editing the start and end points (and maybe setting a nice, clean fade out to clean up the end of the track) and then transferring it back to the DAT tape if you wish. In this way, your DAT recordings can be very professionally edited and presented to a record company or pressing plant.

CREATING AN ALBUM COMPILATION TRACK LIST

You can use the S3200's SONG mode to create an album compilation track list for mastering. Record all your songs into the S3200 (or transfer them from DAT). Now record a 6 second 'dummy' take of silence called SPACE or GAP or something similar. You may now create a song, assigning the recordings in the

order you want them to play back and, in between each one, inserting the silent 'dummy' cue. In this way, you may create several versions of an albums running order for presentation to the record company. Remember, of course, that each track in the list may have its own fade in and fade out set and its own level adjusted to create a perfectly balanced master for the pressing plant.

IMPORTANT NOTE: *Although it is possible to transfer a DAT or CD recording into the S3200 digitally, some restrictions exist that prevent the S3200 transmitting the signal at 48kHz for transfer back to some DAT machines. This is due to hardware restrictions. Most DAT machines, however, do allow you to record digitally at 44.1kHz so there shouldn't be much of a problem. Only those older DAT machines that do not allow digital recording at 44.1kHz cannot be used. In this case, the transfer should be done via the analogue outputs.*

MIXING DOWN THE S3200 TO DISK

One very useful feature of the disk record functions is that you may record sequenced programs in the S3200 directly to its own disk. To do this, connect the left/right mix outputs to the S3200's inputs. Start recording and then start the sequencer - the sequenced programs will be recorded to disk where they may be subsequently edited.

An advance on this is to take the outputs of the S3200 (including individual outputs, of course) through a mixer and into the S3200's inputs. In this way, you may add effects such as reverb and delay and EQ the outputs and you may also, of course, mix in the sounds from other samplers, synths, modules and drum machines to be recorded onto disk.

GLOSSARY

Every new technology invents its own terms to describe new techniques. Digital music is no exception, unfortunately. However, a glossary such as this can help introduce you to the vocabulary and concepts involved. We assume you have a basic knowledge of MIDI, but if the S3200 is your first excursion into MIDI and sampling, we suggest you get hold of an introductory MIDI book, and read it before proceeding much further with your S3200.

ANALOGUE — As opposed to digital. Analogue electronics use continuously varying voltages whereas digital circuitry uses only 'on' and 'offs'.

CROSSFADING — On the S3200, crossfading is the term used to describe the setting of the relative volume of two samples which are played at the same time. For instance velocity crossfading is used to describe the relative balance between two samples played by the same key, when the key is played at different velocities. Positional crossfading refers to relative balance between samples in different keyspans (see EDIT PROGRAM). Additionally, the S3200 allows loop crossfading - the ability to fade samples inside themselves to allow for smooth looping.

CURSOR — On the S3200, the cursor is the highlighted (reversed) part of the display which is moved by the CURSOR keys and indicates the parameter which may be changed by the DATA control and/or the number keypad.

ENVELOPE — The 'shape' of the sound - i.e. percussive, plucked, bowed, blown, etc.. On a synthesizer or sampler, envelope generators allow you to 'shape' the amplitude and tonal dynamics of a sound. Typically, they allow you to emulate the attack, decay sustain and release characteristics of a note.

FIELD — On the S3200, a field is the portion of a page containing a parameter. Only fields (i.e. those portions of a page which may be altered) will be highlighted by the cursor as the CURSOR keys move you through a page.

FILTER — There are two meanings for this on the S3200. One is to remove harmonics out of a sound (see 'harmonics'). The other is to only allow certain MIDI information to pass through the S3200.

HARMONICS — Overtones or extra frequencies in a sound that creates its tonal characteristics. A sound with many harmonics will be bright in tone whilst one with few harmonics will quite mellow. A trumpet, for example, has many harmonics whereas a flute has very few. We can manipulate a sound's harmonic content using filters.

KEY — In this manual, the word 'key' will generally be used to refer to a push-button switch on the front panel. This

	should not be confused with the keys on a musical piano-type keyboard.
KEYGROUP	The term for a collection of a number of parameters of up to four samples, their name, keyspan, filtering, envelopes, etc..
KEYSPAN	On the S3200, a keyspan is the range of the keyboard on which a sample can be played.
LFO	Low Frequency Oscillator - an oscillator which operates at too low a frequency to produce an audible tone, but is used to modulate such parameters as pitch, pan position, etc. The S3200 has a number of LFOs implemented in software.
LOOPING	In sampling, looping refers to the process of taking a portion of a sample and repeating it. The S3200 allows four such loops to be present in a sample, allowing incredibly subtle variations on the basic sampled sound.
MULTISAMPLING	When sampling a sound, replaying it at a radically higher or lower pitch will produce strange and unnatural effects. To overcome this problem, samples should be taken from across the pitch range of an instrument and assigned to different keyspans across the keyboard. This is known as multisampling.
OPERATING SYSTEM	Without a computer program to send signals through the maze of chips and circuitry which make up the hardware of the S3200, the machine would be useless. The program which contains the instructions to respond to MIDI messages, buttons and controllers, and display messages, etc., on screen (as well as to record and play back sounds) is known as the operating system. This is automatically loaded when the S3200 is powered up, either from chips inside the S3200, or, if a disk containing a version of the operating system has been placed in the floppy disk drive, from disk.
OVERTONES	Basically the same as harmonics
PAGE	On the S3200, a page is the set of information and parameters shown at any one time on the display screen. Pages can be entered by pressing the named buttons (EDIT SAMPLE, EDIT PROGRAM, etc.), or the soft keys whose legends are displayed at the bottom of a page.
PARAMETER	A value which can be changed (for instance length, tuning, upper limit of a keyspan) as displayed on the screen of the S3200.
POSITIONAL CROSSFADING	See Crossfading.

PROGRAM

The term for a collection of keygroups which will all be selected together when the program is selected. Different programs can be assigned to different MIDI channels, so that when a sequencer is connected to the S3200, multi-timbral output is possible.

REAL-TIME

As opposed to non real-time - the ability to actually hear the results of your editing without having to wait for the computer (or in this case, the S3200) to do calculations. It is very important for editing to be real-time for successful operation. Most functions on the S3200 are real-time except for such things as copying samples and timestretch. The length of time it takes for these processes depends on the length of the sample.

SAMPLE

Usually in the manual, the word sample will refer to a sound which has been recorded, digitized and edited, and can then be added to a keygroup (you might like to think of it as a "waveform" in analogue synthesizer terms). However, when editing one of these sounds, the length and position inside this sample is also measured n samples. This latter meaning refers to the digitized 'snapshot' image of the sound for one cycle of the sampling process. A sample recorded at 44.1kHz and lasting for exactly one second therefore contains 44,100 samples! It will usually be clear in this manual, however, what meaning of the word sample is meant at any one time - if it is likely to cause confusion, we will refer it as a 'recording'.

SCSI

Short for Small Computer Serial Interface and is a standard communication method between two computers or a computer (i.e the S3200) and a hard disk drive.

SOFT KEY

On the S3200, a key underneath the LCD with no pre-defined function. The current state of the S3200 determines the function, which is displayed on the bottom line of the page.

SPLICING

The process of joining samples to each other (analogous to tape splicing). However, this is much easier electronically than when using razor blades and splicing tape, and many more effects are possible. To take an extreme example, the sound of a string section could be spliced to the sound of a bottle breaking, and the resulting sample then spliced to the reversed sound of the string attack.

VELOCITY CROSSFADING

See Crossfading and Velocity zones.

VELOCITY ZONES

On the S3200, a sample can be programmed to play only when a key is pressed between certain velocities. The range of these is known as a velocity zone. Up to four samples may be assigned in each keygroup, and if desired, each can be assigned to a different velocity zone. In this way, a finger-style electric bass sample

could be assigned to lower velocity zones, and a slap or pull bass to higher ones, with a velocity crossfade added so that there is an intermediate range. The result, when played, will provide a highly expressive bass instrument.

VOLUME

As well as being the output level from the S3200, volume has another meaning - a collection of programs, samples and drum settings which can be stored together on a floppy disk, on a hard disk or in memory. One volume can be stored in memory or on each diskette, and up to 128 volumes can be stored on a hard disk.

S3200 MIDI IMPLEMENTATION CHART

Date: FEB. 1993

Function	...	Transmitted	Recognized	Remarks
Basic Channel	Default Changed	X X	○ 1 ○ 1-16	Without disk Memorized (disk)
Mode	Default Messages Altered	X ************	Mode 3 Mode 1-4 OMNI ON/OFF, P/M X	Without disk Memorized (disk)
Note Number:	True Voice	X ************	21-127 4-127	
Velocity	Note on Note off	X X	○ 9n V=1-127 X 8n V=1-127	 Release Velocity
After-touch	Key's Ch's	X X	X ○	
Pitchbend		X	○	0-24 semitone steps (8-bit resolution)
Control Change	1 2 4 7 64 67	X X X X X X	○ ○ ○ ○ ○ ○	Modulation wheel EWI Breath controller (*1) Foot switch controller (*1) Volume Sustain pedal Soft pedal
Program Change	True No.	X ************	1-128	 by Preset number Value
System Exclusive		○	○	AKAI ID : 47H S3200 ID: 48H (*2)
System Common	: Song position : Song select Tune	X X X	X X X	
System Real time	: Clock : Commands	X X	X X	
Aux Messages	: Local ON/OFF : All Notes OFF : Active Sense : Reset	X X X X	X ○ (123) X X	

Mode 1: OMNI ON, POLY　　Mode 2: OMNI ON, MONO　　○ : Yes
Mode 3 : OMNI OFF< POLY　　Mode 4: OMNI OFF, MONO　　X : No

(*1) Use external Modulation.
(*2) Full details of System Exclusive data formats does not installed software Version 1.0.

SPECIFICATIONS

Model Name	: MIDI Stereo Digital Sampler S3200
Sampling Data format	: 16-bit linear encording
Sampling rates	: 44.1 KHz (20 Hz~20 KHz audio band width)
	22.05 KHz (20 Hz~10 KHz audio band width)
Sampling time	: 1 minute 29 seconds - mono Fs+44.1 KHz
(unexpanded memory)	2 minute 58 seconds - mono Fs+22.05 KHz
	44.56 seconds - stereo Fs+44.1 KHz
	1 minute 29 seconds - stereo Fs=22.05 KHz
Internal Memory	: 8 Mbyte standard, expandable to 32 Mbyte
Polyphony	: 32 Voices
Maximum number of Samples	: 255
Maximum number of Programs	: 254
Filter	: Digital moving low-pass filter (-12 dB/octave with resonant)
Envelope generators	: 3 x digital Envelope generators (2 multi-stage)
L.F.O.	: 2 x Multi Wave Low Frequency Oscillators
Display	: Backlit 320 characters/240 x 640 graphic LCD
Diskette drive	: 3.5" dual density drive (2HD, 2DD)
Connectors	
REC IN	: 2 x XLR (balanced) 2 x 1/4-inch phone (balanced)
STEREO OUT	: 2 x XLR (balanced) 2 x 1/4-inch phone (unbalanced) } 7.5 dBm, 600Ω
ASSIGNABLE OUTS	: 8 x 1/4-inch phone (unbalanced)
HEADPHONES	: 1 X 1/4-inch stereo phone
FOOTSWITCH	: 1 x 1/4-inch phone
MIDI	: 3 X DIN5P (IN, OUT, THRU)
SCSI Interface	: 1 x 50P Amphenol
AES/EBU digital IN/OUT	: 2 x 1/4-inch phone (balanced) and Optical input/output
SMPTE time code IN/OUT	: 2 x 1/4-inch phone (balanced)
REC GAIN	: HI-58 dBm, MID-38 dBm, LO-18 dBm
Power Requirements	: 120VAC 60Hz 40W for U.S.A. and Canada
	220~230VAC 50Hz for Europe (excluding U.K.)
	240VAC 50Hz for U.K. and Australia
Dimensions	: 483W x 133H x 410 (*429) Dmm (EIA 3U size) (*) maximum
Weight	: 10.0 Kg
Accessories	: AC power cable 1
	Sound Library Disks 4
	Operators Manual 1
Optional Accessories	
EXM 3002	: 2 Mbyte memory expansion board
EXM 3008	: 8 Mbyte memory expansion board
BL1000	: 3.5-inch blank diskettes (MF2HD)
HD108	: Hard Disk unit
SMO-P301	: 3.5-inch MO disk drive set for S3200

* Above specifications are subject to change without prior notice.

Version 1.0 March 1993